Soundcraft Vi4™ /Vi6™
User Guide

Soundcraft

by HARMAN

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

This equipment complies with the EMC directive 2004/108/EC
and LVD 2006/95/EC

This product is approved to safety standards
IEC 60065:2001 (Seventh Edition) +A1:2005
EN60065:2002 +A11:2008
UL60065-03
CAN/CSA-E60065-03

And EMC standards
EN55103-1: 1996 (E2)
EN55103-2: 1996 (E2)

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 B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

* Reorient or relocate the receiving antenna
* Increase the separation between the equipment and the receiver
* Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
* Consult the dealer or an experienced radio/TV technician for help

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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.

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Contents

SAFETY NOTICES

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

SAFETY SYMBOL GUIDE

For your own safety and to avoid invalidation of the warranty all text marked with these symbols should be read carefully.

WARNINGS
The lightning flash with arrowhead symbol, is intended to alert the user to the presence of un-insulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.

CAUTIONS
The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

NOTES
Contain important information and useful tips on the operation of your equipment.

HEADPHONES SAFETY WARNING
Contain important information and useful tips on headphone outputs and monitoring levels.

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 equipmentis operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, 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.
This Class A digital apparatus meets the requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la Classe A respecte toutes les exigences du Règlement sur le matériel brouilleur du Canada.

IMPORTANT SAFETY WARNINGS

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 receptacles 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.
- 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.

- This unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Soundcraft dealer.
- Ventilation should not be impeded by covering the ventilation openings with items such as newspapers, table cloths, curtains etc.
- The disconnect device is the mains plug; it must remain accessible so as to be readily operable in use.
- It is recommended that all maintenance and service on the product should be carried out by Soundcraft or its authorised agents. Soundcraft cannot accept any liability whatsoever for any loss or damage caused by service, maintenance or repair by unauthorised personnel.

WORKING SAFELY WITH SOUND

Although your new console will not make any noise until you feed it signals, it has the capability to produce sounds which when monitored through a PA system or headphones can damage hearing over time. The table below is taken from the Occupational Safety & Health Administration directive on Occupational noise exposure (1926.52):

PERMISSABLE NOISE EXPOSURE

DURATION PER DAY, HOURS	SOUND LEVEL dBA SLOW RESPONSE
8	90
6	92
4	95
3	97
2	100
1.5	102
1	105
0.5	110
<0.25	115

Conforming to this directive will minimise the risk of hearing damage caused by long listening periods. A simple rule to follow is the longer you listen the lower the average volume should be.
Please take care when working with your audio - if you are manipulating controls which you don't understand (which we all do when we are learning), make sure your monitors are turned down. Remember that your ears are the most important tool of your trade, look after them, and they will look after you.
Most importantly - don't be afraid to experiment to find out how each parameter affects the sound - this will extend your creativity and help you to get the best results.

Recommended headphone impedance is 50-600 ohms.

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; and
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, air-conditioning 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.

Soundcraft Vi Series™ FEATURES AND SPECIFICATIONS

Audio Channels

Max number of simultaneous mixing channels
Soundcraft Vi6 : 96 mono inputs into 35 mix busses. (3 DSP cards fitted)
Soundcraft Vi4 : 96 mono inputs into 35 mix busses. (3 DSP cards fitted)
: 64 mono inputs into 35 mix busses (2 DSP cards fitted)
Soundcraft Vi2 : 96 mono inputs into 35 mix busses. (3 DSP cards fitted)
Pairs of mono inputs can be linked to create stereo channels.

Insert points
24 insert send/return pairs can be configured (using available I/O) and assigned to any of the 96 inputs or 35 output channels

Direct Outputs
All 96 input channels can have direct outputs in addition to their internal bus routing, assuming sufficient I/O is available (eg via 64ch optical MADI card, see below)

Busses
32 Grp/Aux/Matrix*, plus main LCR Mix and LR Solo busses.
* a maximum of 16 matrix outputs can be configured.

I/O Capability

The following I/O is available and can be patched to any channel input, direct output, bus output or insert point as required:

Local Rack Inputs
- 16 analogue line inputs
- 3 analogue mic/line inputs
- 1 Talkback Mic input (mounted on control surface – 2 parallel sockets front/rear)
- 8 pairs of AES/EBU inputs (=16 channels)
- 64ch MADI In via optical SC connectors

Local Rack Outputs
- 16 analogue line outputs
- 8 pairs of AES/EBU outputs (= 16 channels)
- LCR Local monitor A analogue line outputs
- LR Local Monitor B analogue line outputs
- TB line output
- 64ch MADI Out via optical SC connectors

Stagebox Inputs (max configuration)
- 64 analogue mic/line inputs (with remote gain control, PAD, 48V and pre-A-D 80Hz HPF)

Stagebox Outputs (max configuration)
- 32 analogue line outputs (max configuration)

Miscellaneous

Connection from local rack to stagebox

Standard fit: Cat 5e Neutrik Etherflex cable ZNK CT2672601.
Optional: Fibre Optical interface card with 150 or 200m cable (additional cost).

Max distance, local rack to stagebox:

80m using flexible reel-mounted Cat5 cable (Neutrik Etherflex only, part number ZNK CT2672601)
130m using Cat7 permanent installation cable (Amp Netconnect 600MHz PiMF, part no. 57893-x).
1500m using a single run of multimode 50/125 optical fibre.
600m using 3 X 200m reels of multimode 50/125 optical fibre joined in series.

GPIO facility

- 16 GPIO inputs and outputs on the local rack
- 8 GPIO inputs and outputs on the stagebox (All outputs are relay contact closure)

MIDI

- 1 MIDI Input and 2 MIDI Outputs on rear of control surface.

Channel Processing

Inputs

- Analogue gain (remote control of stagebox or local mic preamp)
- Digital Gain Trim (+18/-36dB)
- Delay (0-100ms)
- HPF, LPF (variable 20-600Hz and 1-20kHz)
- 4-band fully parametric EQ, shelf mode on HF/LF.
- Compressor (variable threshold, attack, release, ratio, makeup gain with ‘auto’ mode)
- Limiter (variable threshold, attack, release)
- Gate or De-Esser. Gate switchable to ducker.
- Insert point for external processing.
- Pan – LR or LCR switchable.
- Direct Output, patchable to any I/O and with selectable tap-off point.

Outputs

- HPF (variable 20-600Hz)
- 4-band fully parametric EQ, shelf mode on HF/LF.
- Compressor
- Limiter
- Delay (0-1sec)
- Insert point for external processing.
- Pan (Output bus to LCR) – LR or LCR switchable.
- Bus Feed feature – allows switched routing of one bus to another.
- Graphic EQ 1/3-octave (with FX Card)
- Assignable Lexicon Multi-FX processors x8 (with FX Card)

Control Surface

Inputs
Soundcraft Vi6: 32 input faders, switchable in 3 fixed layers to access 96 inputs (3 DSP cards fitted).

Soundcraft Vi4: 24 input faders, switchable in 2 fixed layers to access 64 inputs (2 DSP cards fitted).
: 24 input faders, switchable in 3 fixed layers and meter screens to access 96 inputs (3 DSP cards fitted).

Soundcraft Vi2: 8 input faders, switchable in 3 fixed layers, 3 user layers and meter screens to access 96 inputs.

Vistonics™ II channel strip interface x 4 (3 on Vi4), each Vistonics™ controls 8 input channels.
The Vistonics™ II interface contains 16 rotary encoders and switches, and a touch screen.

Fader tray contains motorised faders, Mute, Solo, Isolate and F (user defined) switches, plus one assignable rotary encoder with LED display ring. This encoder is globally assignable to Gain, Pan, Gate Threshold, or one of 2 user-definable parameters.

Input level and gain reduction meters are located above each fader.

Input faders can be assigned to the 16 VCA (control group) masters and/or 4 Mute Groups.

Input faders can be switched to control all 32 Grp/Aux/Matrix Outputs, or can control an individual Aux send mix, using the switchable 'Follow Solo' function. Soundcraft FaderGlow™ clearly indicates using colours when faders are not controlling inputs.

Outputs
8 assignable Output faders, plus dedicated LR and C Master faders, plus 16 assignable rotary Output faders. Output faders are colour-coded using Soundcraft FaderGlow.

Output faders can be assigned to the 16 VCA (control group) masters and/or 4 Mute Groups.

Single Vistonics II interface for Output processing control, also functions as complete meter overview display for all Inputs & Outputs, plus snapshot Cue List and diagnostics info display.

Misc
Gang mode for temporary linking of any number of channels or outputs for quick adjustment and setup
Controls for Mute Group and VCA Group assignment.
Controls for assignment of Vistonics™ rows to bus sends (when channel parameters are not assigned to Vistonics).
Snapshot automation controls.
Talkback & Oscillator controls.
Controls for Monitor Output level, phones level and Solo Trim and blend level.

CONSOLE OVERVIEW

Bays

The desk is based on 1, 3 or 4 bays and 1 MASTER bay:

Soundcraft Vi2:
* 1 Input Bay

Soundcraft Vi4:
* 3 INPUT bays

Soundcraft Vi6:
* 4 INPUT bays

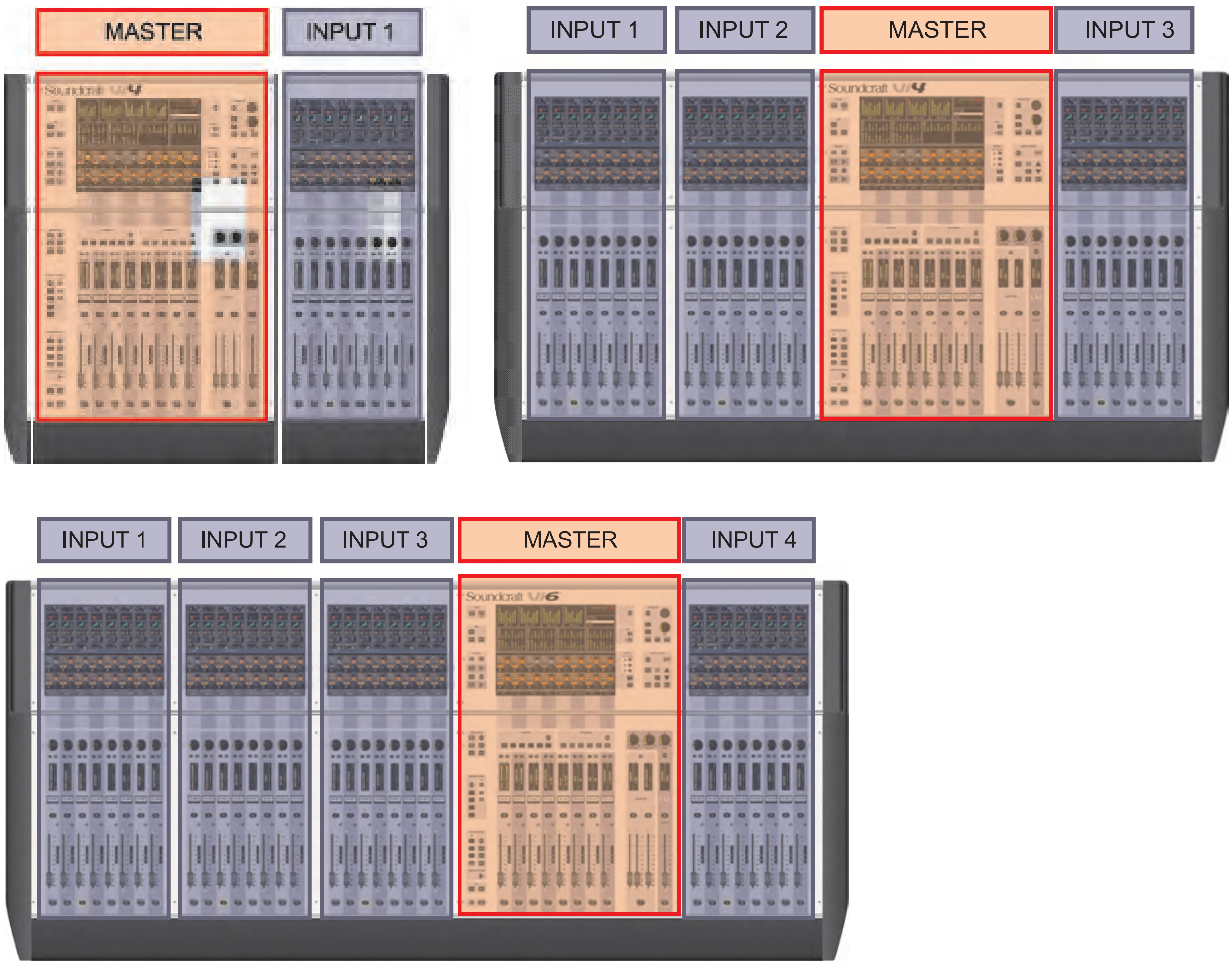


Figure 1-1. Console Bays.

* Each Input bay contains 8 complete Fader strips with full state overview, giving in total 32 directly accessible Input Strips.

* The Master bay contains 8 Output Fader strips, the 3 Masters and 16 Output encoders that give a total of 27 Output levels that can be directly controlled without changing Layers. General Functions like Snapshot, Monitoring, TB & OSC and so on are also located on the Master bay.

Layers

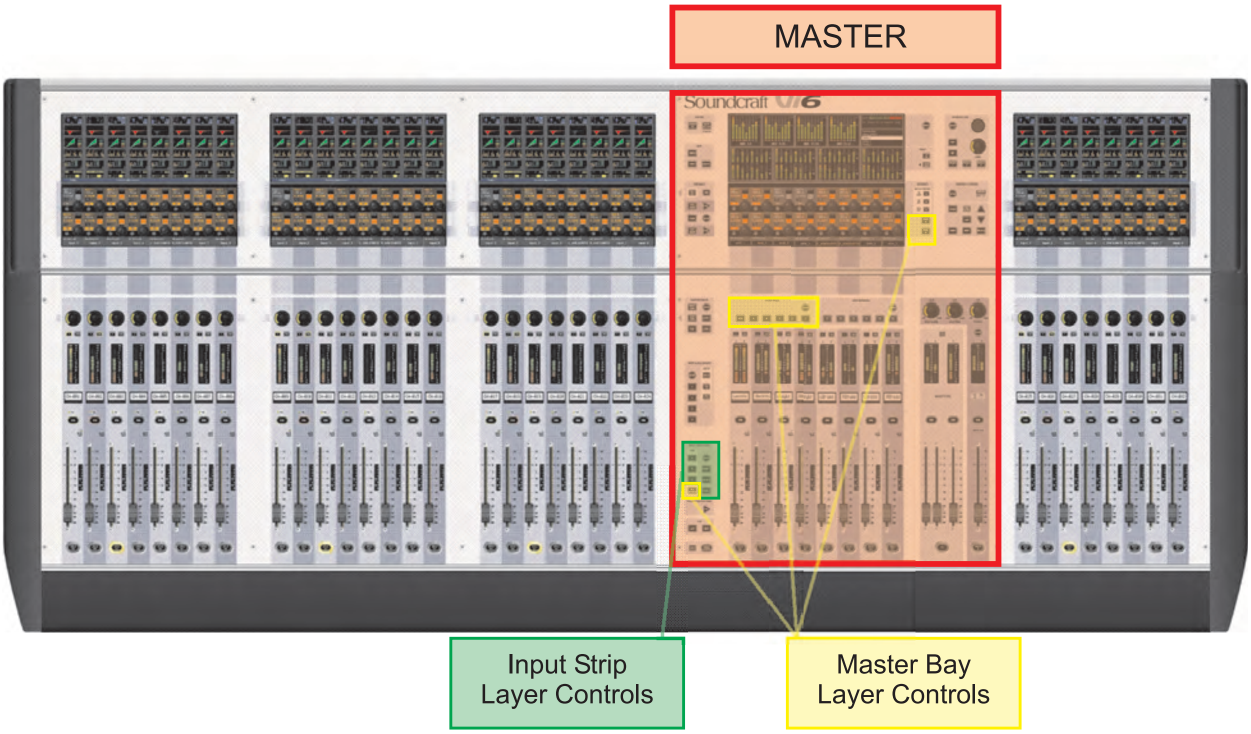


Figure 1-2. Layer Controls.

The console is able to control up to 96 inputs and 35 mix busses via its 32 input strips (24 on Vi4, 8 on Vi2), 8 bus master strips and the LCR masters. To do this the console has a number of layers which the user accesses via the layer controls shown above.

Full details are given in chapter 8 of this manual.

Encoders

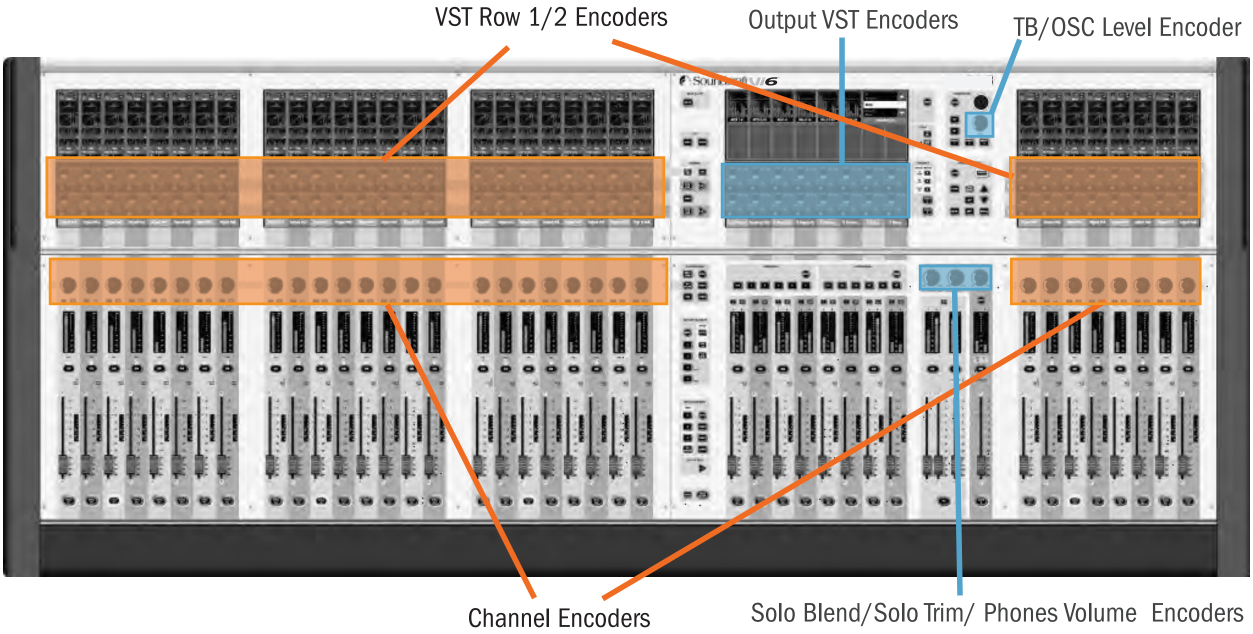


Figure 1-3. Encoders.

* The Vistonics™II **row 1 and 2 encoders**, including switches, are used in different modes, in which they can change in order to show various functions in a context-sensitive way.

* In normal operation they act as Input channel related controls.

* The **Channel encoders** are assigned with Input channel related functions.

* The Vistonics™ II **output encoders** are normally used as Output faders and are also context sensitive.

* There are four panel-mounted encoders with LED rings: the TB/OSC Level Control encoder, and the Sold Blend, Solo Trim & Phones Volume encoders. These are dedicated to their respective functions.

A detailed explanation of encoder use is given in chapter 7 of this manual.

Master Audio Functions

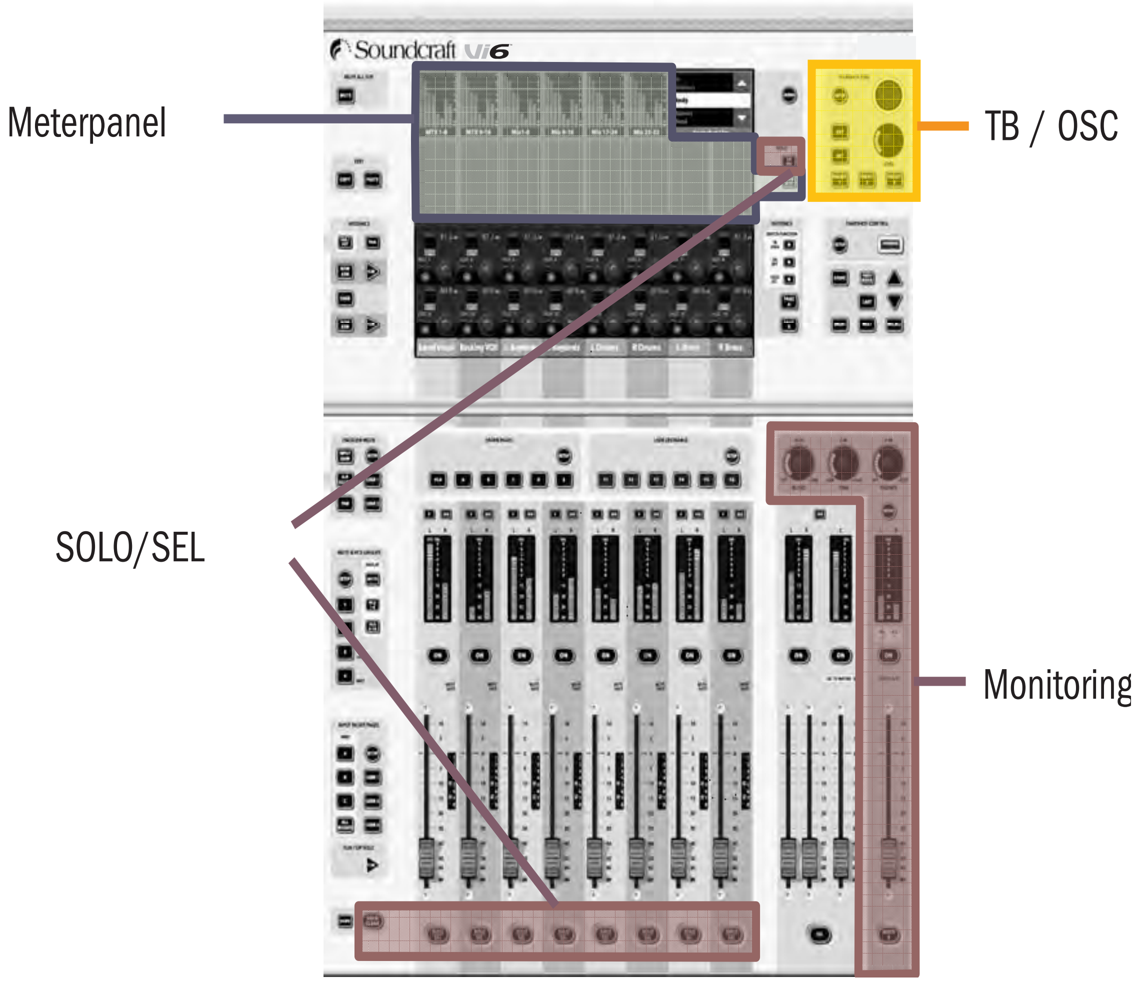


Figure 1-4. Master Audio Functions.

* The Monitoring system contains the functionality to listen to and monitor the audio signal at several points in the console.

* TB/OSC system contains the Talkback functionality and the oscillator settings.

* The Meter panel provides a full overview of all Input and Output levels.

Master Control

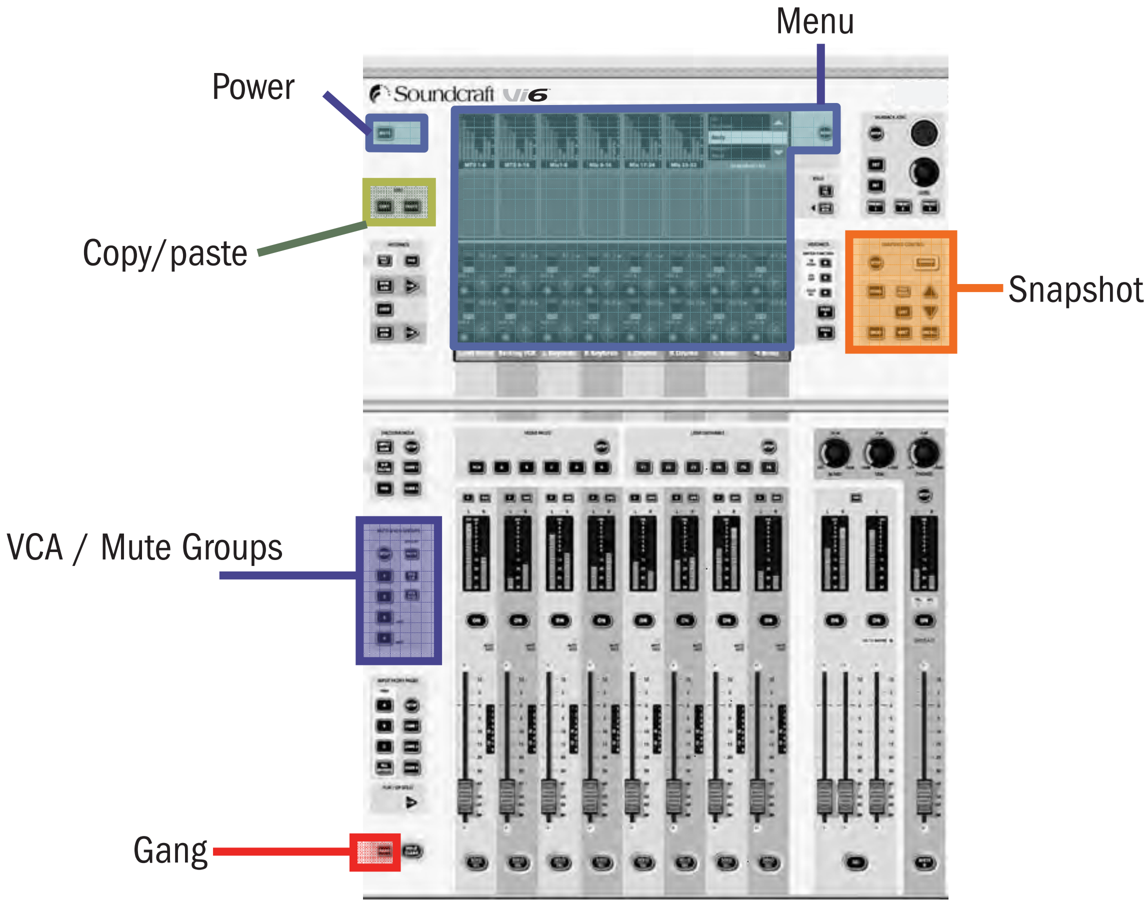


Figure 1-5. Master Control.

* VCA/Mute Groups: this functional block contains the VCA (control groups in VCA style) and Mute Group functions.

* Snapshot allows the console's automated settings to be saved and recalled.

* Menu opens the Menu page where central configurations can be done.

* Copy / paste functionality can be used in different modes and speeds up repetative tasks..

* Gang is a superb feature that links channels functions together for temporary changes.

* [Power] switches the Desk on and off, while [MUTE ALL] Outputs is helpful in emergency situations.

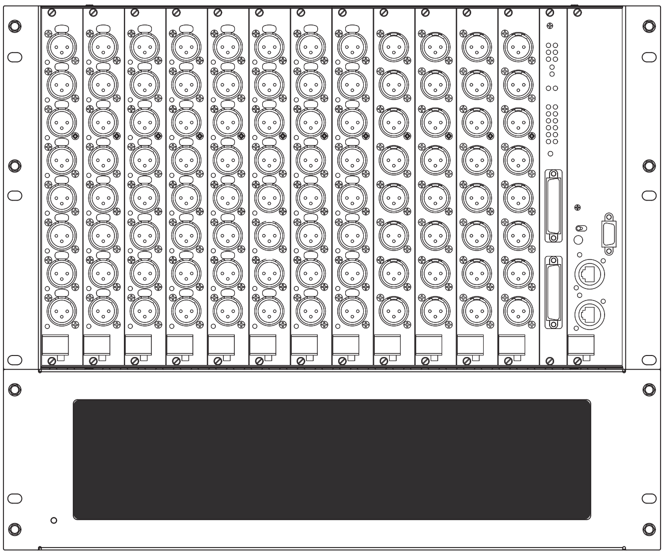
SYSTEM COMPONENTS

SYSTEM HARDWARE OVERVIEW

Stage Box

6U Stage Box

2U Fan Unit

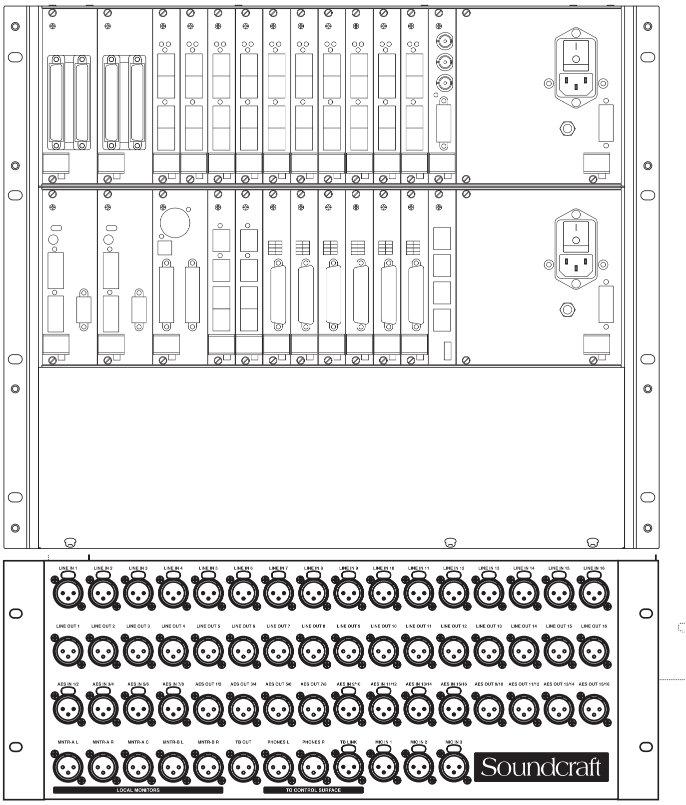


Local Rack

6U Local Rack

3U Fan/Filter Unit

4U Breakout Box



Console Surface

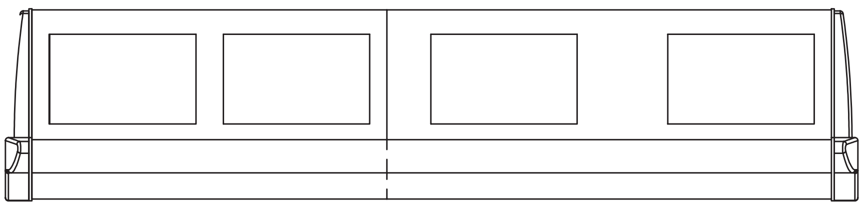


Figure 2-1. The System's Three Component Parts.

NOTE ON INSTALLATION OF THE COOLING FANS/FILTERS

The Stage Box and Local Rack may be ordered already fitted into flight cases, in which case the cooling fans/filters will already be located correctly as shown in the diagrams below.

If the system has been ordered without flightcases, in order that it can be permanently installed, please ensure that the cooling fans/filters are located as shown below.

Local Rack

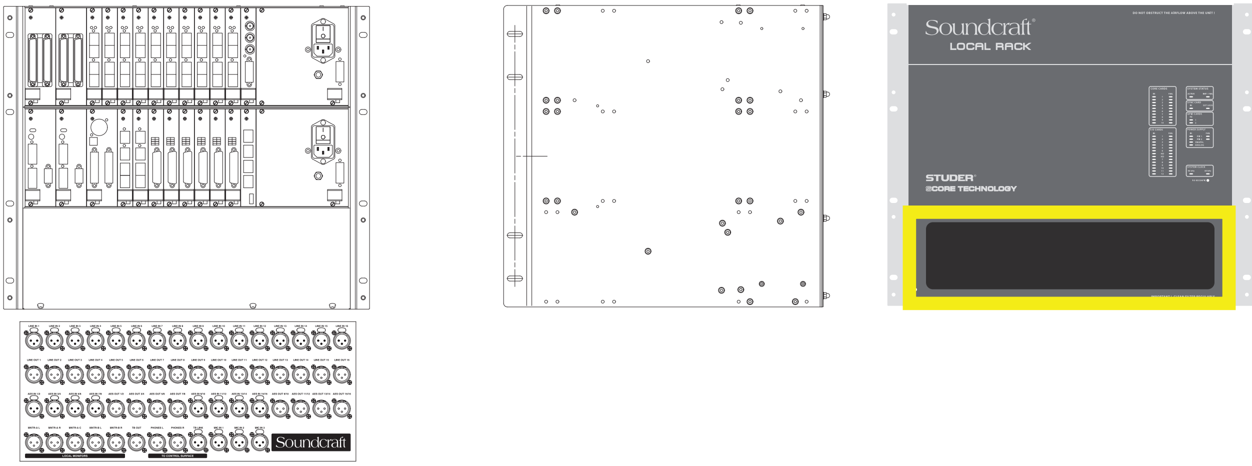


Figure 2-2a: Layout Of Local Rack.

Stage Box

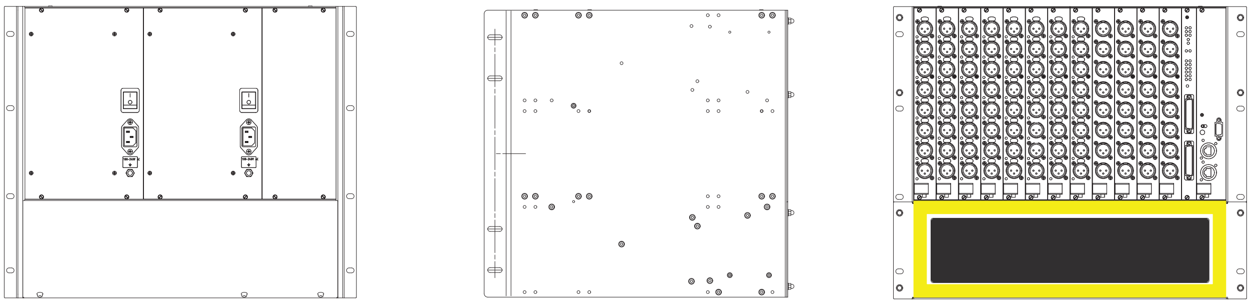


Figure 2-2b: Layout Of Stage Box.

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

CLEAN FILTERS REGULARLY!

The filters are outlined in yellow in the figures above.

THE CONSOLE REAR CONNECTORS

Mains Power Supply Inlet

The mains input is via an IEC connector, with an associated switch, as shown below. This feeds power to the primary PSU.

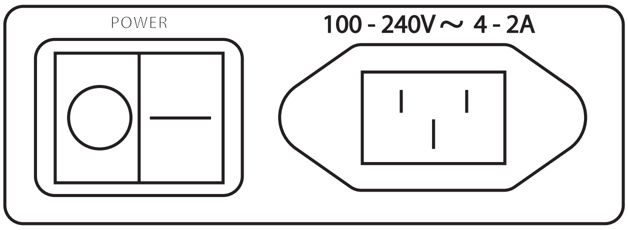


Figure 2-3a: Main Primary Supply Inlet.

An optional redundant primary supply may also be fitted. Its inlet connector is as shown below.

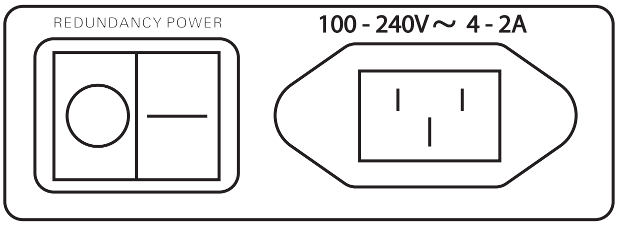


Figure 2-3b: Redundant Primary Supply Inlet.

Audio And Data Rearcon

MIDI Connectors

These are provided by the usual 5-pin DIN connectors.

HiQNet™ Connector

This is an XLR-housed EtherCon connector.

USB Connectors

Use one of these to connect a PC-type keyboard. There is another USB connector on the front panel. A memory stick can be used with either of these two free connectors.

Talkback Mic

This connector is a parallel connection to the front-panel talkback mic connector, and has an associated 48V switch for use if the microphone used requires phantom power.

TB Link, Phones L & R and Control Data

This group of four sockets (3 XLRs and an XLR-housed EtherCon connector) are used to link the console's control surface to the Local Rack. The cable to do this (part number RL0267-01) is supplied with the system. The function of each of the connectors in this cable is marked with a cable sleeve.

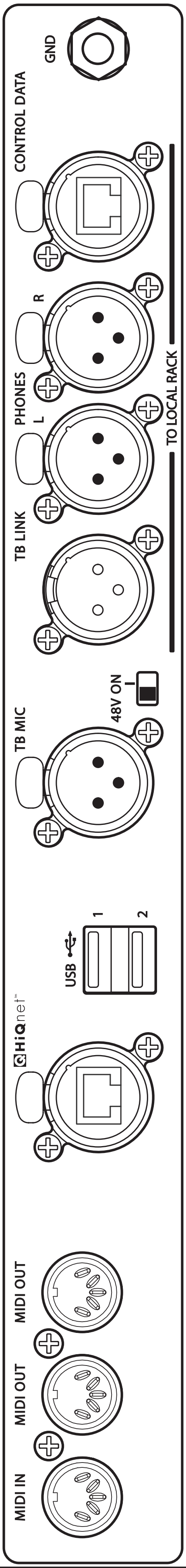


Figure 2-4: Console Rearcon Panel.

LOCAL RACK

Local Rack Description

The Local Rack is the audio 'brain' of the Soundcraft Vi Series™, it contains the DSP mixing processor and the local I/O connections.

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. 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 two DSP Pro cards and the FX Card (when fitted). These cards actually process the audio. The audio input and output connections to the DSP cards are made via 4 short CAT 5 patch cables, which connect the DSP cards with the I/O rack in the lower 3U section of the core.

Front Panel

The front panel contains the Status LED array.

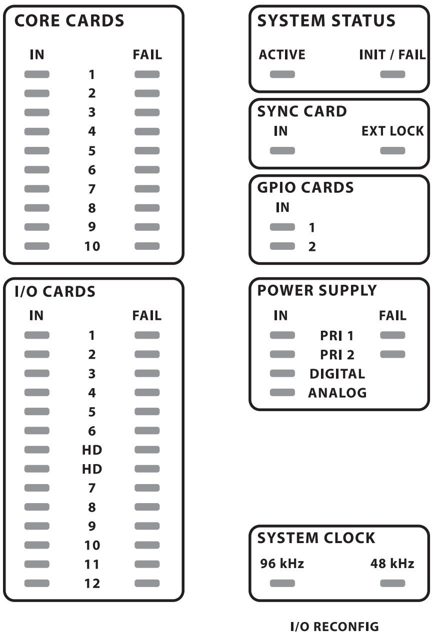


Figure 2-5: Status LEDs.

These LEDs give an at-a-glance indication that the Local Rack is functioning correctly, or that an error condition has occured. 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.

Local Rack Rear

The rear of the Local Rack gives access to the cards in the card frame, and to the XLR breakout box.

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 Local Rack can hold up to two power supplies, providing seamless redundancy for those that require it. The rack fan control connection is also provided by one of the power supplies.

XLR Breakout Box

This provides connectors for analogue and AES/EBU audio, and interfaces to the Line In cards, Line Out cards, Mic card and AES/EBU Card within the Local Rack card frame. All the connectors on the Breakout Box are of the XLR type.

It is possible to order the console without this part, in this case the user will have to provide suitable cabling and connectors to interface with the appropriate Local Rack cards. A complete pin list is given later in this chapter.

The Breakout Box connector 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.

Audio I/O Cards

The following cards are supported.

1 X Mic/Line Input card providing four electronically balanced Mic/Line Input channels, each with digitally-controlled analogue gain, a 80Hz low-cut filter, and phantom power.
2 X Line Input cards, each providing eight line input channels.
3 X Line Output cards, each providing eight line output channels.
1 X AES I/O card, providing 16 AES input channels and 16 AES output channels.
1 X MADI Stage Box interface card (normally Cat 5, but it can optionally be replaced by an optical MADI card).
1 X Optical MADI card, providing 64 MADI input and 64 MADI output channels.

See page 2-10 for more details.

Local Rack Rear View

The rear view is shown below. Note that the last remaining connector from the console surface (the RJ45 ethernet connector) goes into the top socket on the bridge card.

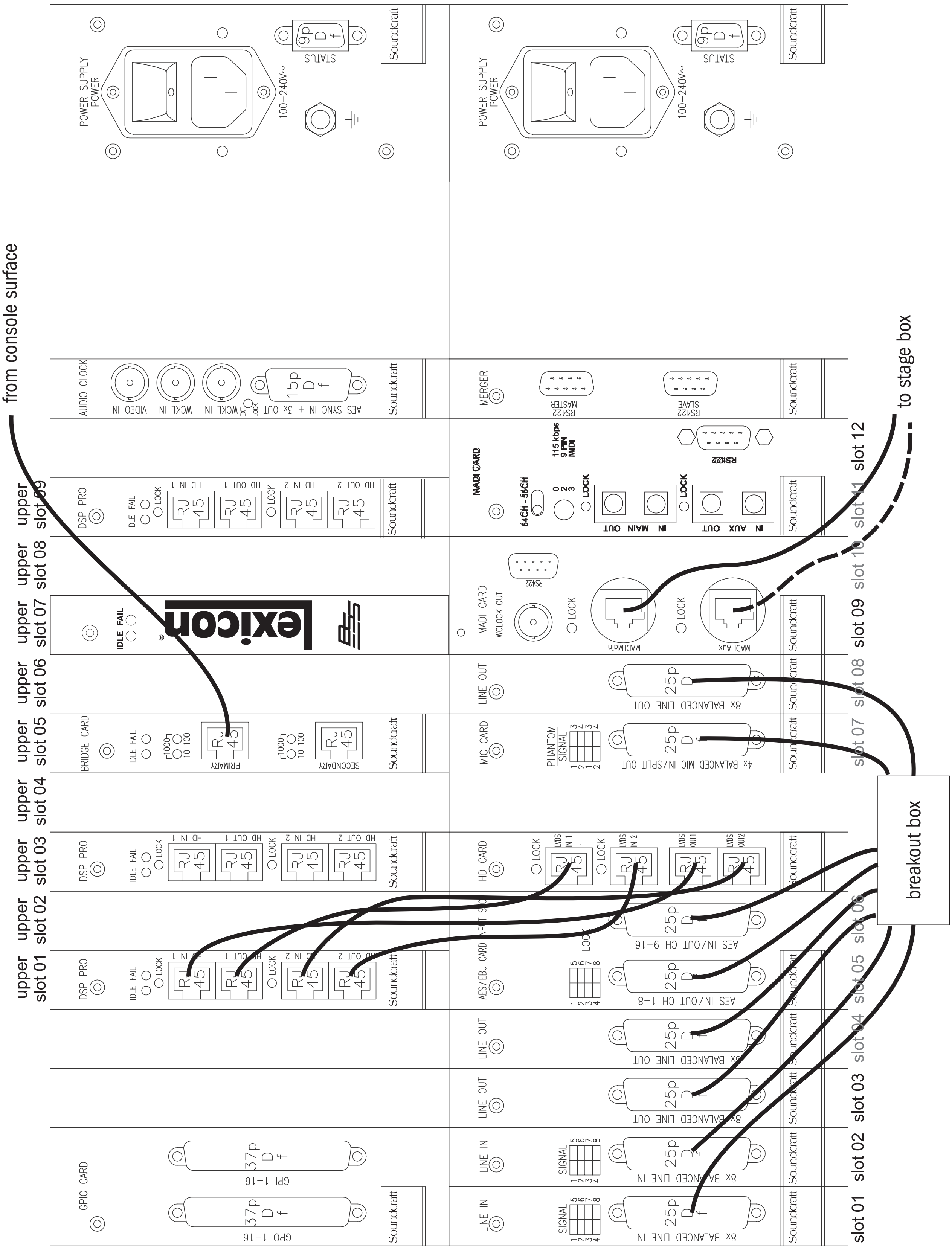
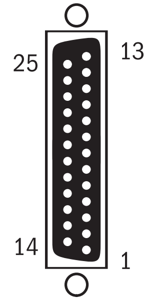


Figure 2-6: Local Rack Rear View.

Local Rack Audio Pinouts

ALL CONNECTORS ARE
25-PIN D-TYPE FEMALE



SLOT 1 LINE INPUTS

Pin	Signal	Pin	Signal
		13	no connection
25	LINE IN 1 screen	12	LINE IN 1 signal-
24	LINE IN 1 signal+	11	LINE IN 2 screen
23	LINE IN 2 signal-	10	LINE IN 2 signal+
22	LINE IN 3 screen	9	LINE IN 3 signal-
21	LINE IN 3 signal+	8	LINE IN 4 screen
20	LINE IN 4 signal-	7	LINE IN 4 signal+
19	LINE IN 5 screen	6	LINE IN 5 signal-
18	LINE IN 5 signal+	5	LINE IN 6 screen
17	LINE IN 6 signal-	4	LINE IN 6 signal+
16	LINE IN 7 screen	3	LINE IN 7 signal-
15	LINE IN 7 signal+	2	LINE IN 8 screen
14	LINE IN 8 signal-	1	LINE IN 8 signal+

SLOT 2 LINE INPUTS

Pin	Signal	Pin	Signal
		13	no connection
25	LINE IN 9 screen	12	LINE IN 9 signal-
24	LINE IN 9 signal+	11	LINE IN 10 screen
23	LINE IN 10 signal-	10	LINE IN 10 signal+
22	LINE IN 11 screen	9	LINE IN 11 signal-
21	LINE IN 11 signal+	8	LINE IN 12 screen
20	LINE IN 12 signal-	7	LINE IN 12 signal+
19	LINE IN 13 screen	6	LINE IN 13 signal-
18	LINE IN 13 signal+	5	LINE IN 14 screen
17	LINE IN 14 signal-	4	LINE IN 14 signal+
16	LINE IN 15 screen	3	LINE IN 15 signal-
15	LINE IN 15 signal+	2	LINE IN 16 screen
14	LINE IN 16 signal-	1	LINE IN 16 signal+

SLOT 3 LINE OUTPUTS

Pin	Signal	Pin	Signal
		13	no connection
25	LINE OUT 1 screen	12	LINE OUT 1 signal-
24	LINE OUT 1 signal+	11	LINE OUT 2 screen
23	LINE OUT 2 signal-	10	LINE OUT 2 signal+
22	LINE OUT 3 screen	9	LINE OUT 3 signal-
21	LINE OUT 3 signal+	8	LINE OUT 4 screen
20	LINE OUT 4 signal-	7	LINE OUT 4 signal+
19	LINE OUT 5 screen	6	LINE OUT 5 signal-
18	LINE OUT 5 signal+	5	LINE OUT 6 screen
17	LINE OUT 6 signal-	4	LINE OUT 6 signal+
16	LINE OUT 7 screen	3	LINE OUT 7 signal-
15	LINE OUT 7 signal+	2	LINE OUT 8 screen
14	LINE OUT 8 signal-	1	LINE OUT 8 signal+

SLOT 4 LINE OUTPUTS

Pin	Signal	Pin	Signal
		13	no connection
25	LINE OUT 9 screen	12	LINE OUT 9 signal-
24	LINE OUT 9 signal+	11	LINE OUT 10 screen
23	LINE OUT 10 signal-	10	LINE OUT 10 signal+
22	LINE OUT 11 screen	9	LINE OUT 11 signal-
21	LINE OUT 11 signal+	8	LINE OUT 12 screen
20	LINE OUT 12 signal-	7	LINE OUT 12 signal+
19	LINE OUT 13 screen	6	LINE OUT 13 signal-
18	LINE OUT 13 signal+	5	LINE OUT 14 screen
17	LINE OUT 14 signal-	4	LINE OUT 14 signal+
16	LINE OUT 15 screen	3	LINE OUT 15 signal-
15	LINE OUT 15 signal+	2	LINE OUT 16 screen
14	LINE OUT 16 signal-	1	LINE OUT 16 signal+

SLOT 5 AES IN/OUT CHANNELS 1-8

Pin	Signal	Pin	Signal
		13	no connection
25	CH1/2 IN screen	12	CH1/2 IN signal-
24	CH1/2 IN signal+	11	CH3/4 IN screen
23	CH3/4 IN signal-	10	CH3/4 IN signal+
22	CH5/6 IN screen	9	CH5/6 IN signal-
21	CH 5/6 IN signal+	8	CH7/8 IN screen
20	CH7/8 IN signal-	7	CH7/8 IN signal+
19	CH1/2 OUT screen	6	CH1/2 OUT signal-
18	CH1/2 OUT signal+	5	CH3/4 OUT screen
17	CH3/4 OUT signal-	4	CH3/4 OUT signal+
16	CH5/6 OUT screen	3	CH5/6 OUT signal-
15	CH5/6 OUT signal+	2	CH7/8 OUT screen
14	CH7/ 8 OUT signal-	1	CH7/8 OUT signal+

SLOT 6 AES IN/OUT CHANNELS 9-16

Pin	Signal	Pin	Signal
		13	no connection
25	CH9/10 IN screen	12	CH910 IN signal-
24	CH9/10 IN signal+	11	CH11/12 IN screen
23	CH11/12 IN signal-	10	CH11/12 IN signal+
22	CH13/14 IN screen	9	CH13/14 IN signal-
21	CH13/14 IN signal+	8	CH15/16 IN screen
20	CH15/16 IN signal-	7	CH15/16 IN signal+
19	CH9/10 OUT screen	6	CH9/10 OUT signal-
18	CH9/10 OUT signal+	5	CH11/12 OUT screen
17	CH11/12 OUT signal-	4	CH11/12 OUT signal+
16	CH13/14 OUT screen	3	CH13/14 OUT signal-
15	CH13/14 OUT signal+	2	CH15/16 OUT screen
14	CH15/16 OUT signal-	1	CH15/16 OUT signal+

SLOT 7 MIC CARD

Pin	Signal	Pin	Signal
		13	no connection
25	TB LINK screen	12	TB LINK signal-
24	TB LINK signal+	11	MIC IN 1 screen
23	MIC IN 1 signal-	10	MIC IN 1 signal+
22	MIC IN 2 screen	9	MIC IN 2 signal-
21	MIC IN 2 signal+	8	MIC IN 3 screen
20	MIC IN 3 signal-	7	MIC IN 3 signal+
19	not used	6	not used
18	not used	5	not used
17	not used	4	not used
16	not used	3	not used
15	not used	2	not used
14	not used	1	not used

SLOT 8 LINE OUTPUTS

Pin	Signal	Pin	Signal
		13	no connection
25	MNTR A L screen	12	MNTR A L signal-
24	MNTR A L signal+	11	MNTR A R screen
23	MNTR A R signal-	10	MNTR A R signal+
22	MNTR A C screen	9	MNTR A C signal-
21	MNTR A C signal+	8	MNTR B L screen
20	MNTR B L signal-	7	MNTR B L signal+
19	MNTR B R screen	6	MNTR B R signal-
18	MNTR B R signal+	5	TB OUT screen
17	TB OUT signal-	4	TB OUT signal+
16	PHONES L screen	3	PHONES L signal-
15	PHONES L signal+	2	PHONES R screen
14	PHONES R signal-	1	PHONES R signal+

Card Function Overview

The local rack card frame contains the following components.

(Top Row)

GPIO Card with Relay Contacts

Provides a second, independent set of GPIO, which will be controlled by the GPIO page of the Control Surface. The inputs are opto-isolated and the outputs are SPST relay contacts. For a pin-out list see chapter 16.

Bridge Card

Provides the connection from the Local Rack to the Control Surface. Houses a QNX embedded processing system which controls all the audio processing in the rack.

DSP Pro (2 off)

Houses 12 x Hammerhead SHARC chips, which process all audio for the system.

Audio Clock

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.

(Bottom Row)

Line In

Each card provides 8 x transformer isolated analogue inputs at line level. The card has a signal present indicator for each input. Analogue to digital conversion is handled by a 24 bit ADC.

Line Out

Each card provides 8 x transformer isolated analogue outputs at line level. Card provides 8 x DAC. Clock received from backplane. The 3rd card is used for the analogue monitor outputs, and to send phones L & R to the control surface.

AES/EBU Card Input SRC

Card handles 8 x AES input and 8 x AES output. Inputs are sample rate converted so no external clock is required. Inputs have signal present indication. Sample Rate Conversion (SRC) can be disabled using internal jumpers if required.

HD Card

Communicates with the first DSP Pro on the top row of cards. It passes digital audio along with control data and status information about the cards loaded into the frame. Each pair of links can handle up to 96 channels in both directions. 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.

MADI Card 1

This card is used to connect the Local Rack to the Stagebox. This card passes up to 64 channels of audio in each direction between the stagebox and local rack, normally configured as 64 ins and 32 outs. The card also sends control data to the stagebox, as well as receiving information about the status of the stagebox cards. The second input on the card can be used to increase sample rate to 96kHz (this will also require additional DSP and isn't available yet) or provide redundancy. It also carries clock and control data. The 9-pin D-type connector can carry independent RS422 serial data from local rack to stagebox. Lock LED indicates clock lock between local rack and stagebox. Wordclock output can be used as a W/C source when connecting two consoles to one Stagebox.

MADI Card 2

A second MADI card is fitted to the right of the first card. This second card is used for connection to an external MADI device, such as a MADI recorder, and will always be fitted with optical connectors. It is possible to select either 56ch or 64ch MADI mode, depending on the connected device.

Merger Card

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

Breakout Box

The breakout box provides connections for 16 line inputs, 16 line outputs, 16 AES/EBU input channels, 16 AES/EBU output channels, 5 local monitor outputs, a talkback output, 3 mic inputs, and 3 connectors for the link cable to the control surface.

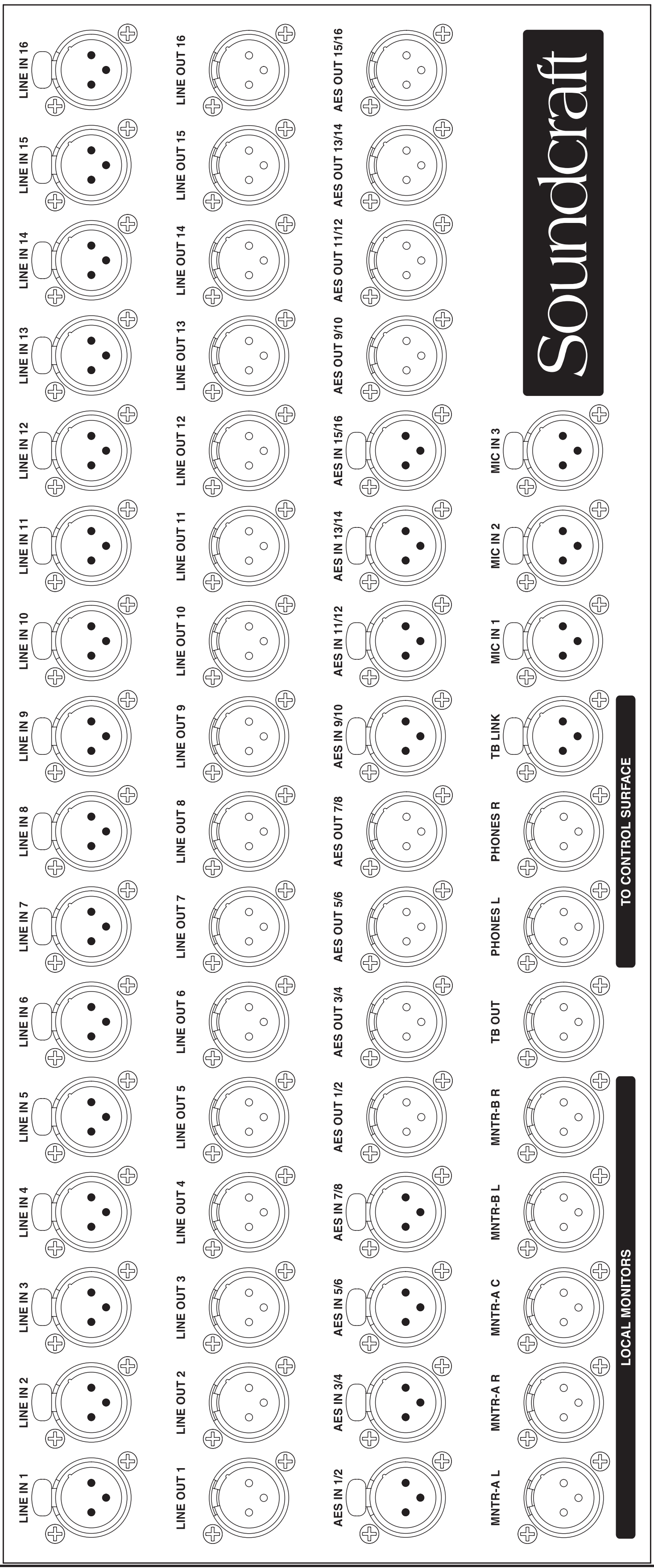


Figure 2-7: XLR Breakout Box.

Cat 5 Breakout Panel

Note that this panel was not fitted to early models.

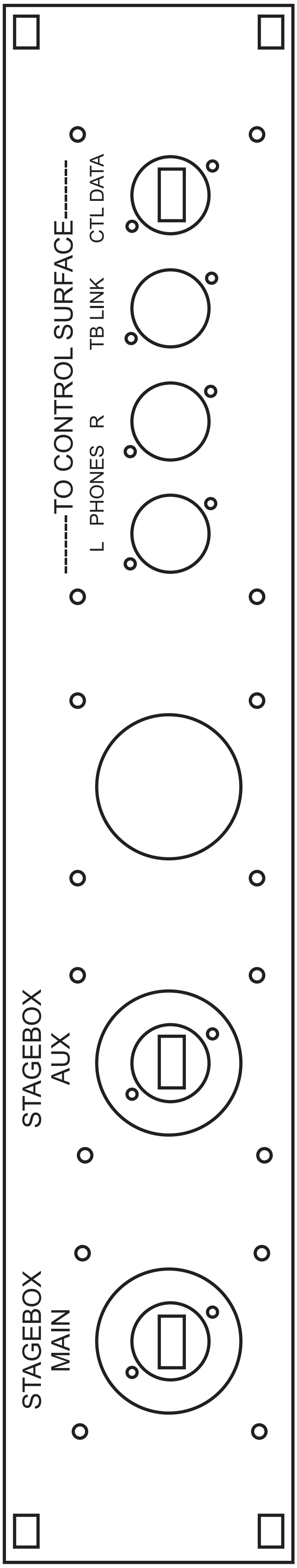


Figure 2-8: Cat 5 Breakout Panel.

STAGE BOX

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 the ventilation monitoring connector can be found on the rear panel.

Stage Box Description

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 can hold up to two power supplies, providing redundancy for those that require it.

Audio I/O Cards
The following cards are supported.

6 X Mic/Line Input cards each providing eight electronically balanced Mic/Line Input channels, each with digitally-controlled analogue gain, a 20dB pad, a 80Hz low-cut filter, and phantom power.
3 X Line Output cards, each providing eight line output channels.
Optional Cards
AES Input card, providing 8 AES input channels (replaces 1 Mic/Line Input card).
AES Output card, providing 8 AES output channels (replaces 1 Mic/Line Output card).

Card Function Overview

Input card
Input cards handle 8 x mic amp, phantom power, pad, analogue low pass filter, phase reverse and A to D. The card has an internal ID, which indicates whether it is input or output and analogue or digital. This means that the system can automatically recognise if the card configuration has been changed.

Output Card
Output cards handle 8 x D to A. The card has a set of relays, which will mute the outputs if the power fails. The module type is identified by the internal ID of the module.

Normally 3 output cards are fitted, giving 24 outputs, however, more cards, up to a maximum of 8 cards giving 64 outputs, can be fitted if input cards are removed.

LED/GPIO/Status card.
Handles GPIO, which is controlled remotely from the Control Surface. The inputs are on opto-isolators and the outputs are open collector transistor outputs. The card also has status indicators for power rails, clock status and IO, and a RECONFIG button which must be pressed if the card configuration has been changed.

MADI HD link card
This card provides audio and control connection with the Local Rack via MADI. The corresponding MADI card in the Local Rack transmits the clock for the Stagebox down the MADI stream. The second input on the card can be used to increase the sample rate to 96KHz, provide a redundant connection to the Local rack or to connect to a second system if two consoles are to be used for a monitor/FOH configuration. The MADI card indicates its clock status using the lock LED on the card. An RS422 link output is also fitted, allowing RS422 data to be transmitted via a 'pipeline' within the MADI stream from a corresponding port on the Local Rack to allow remote RS422 control. This RS422 port can also be used to control the mic preamp gain in the Stagebox if it is in stand-alone mode or digital snake mode.

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.

It is very important that this switch is set correctly!

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.

NOTE: Do NOT operate with the switch in 'RED' mode when only one cable is being used. Although this may appear to work, audio may not be restored if the cable is temporarily unplugged.

Front Panel

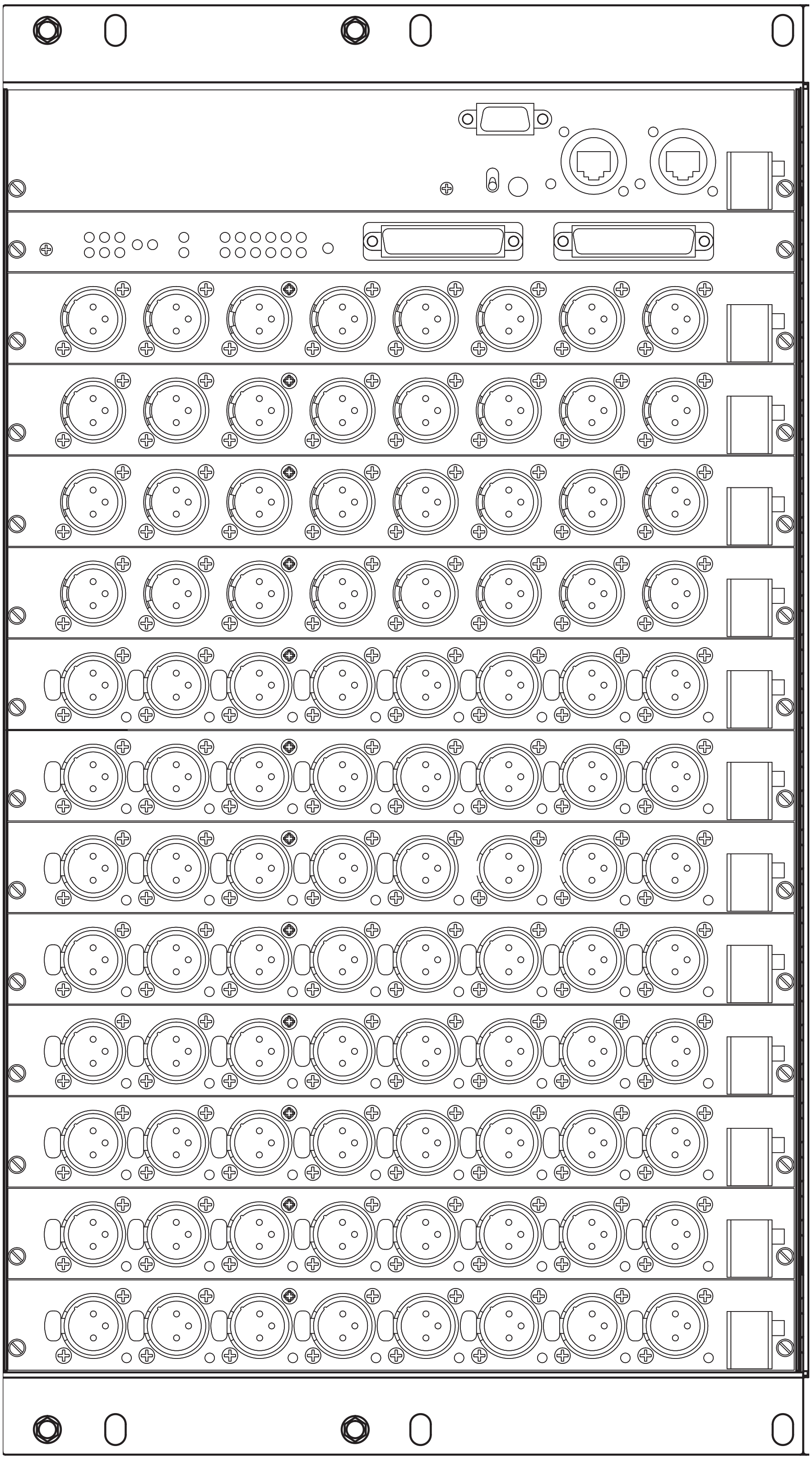


Figure 2-9: Stage Box Front Panel.

CONNECTING THE PARTS OF THE SYSTEM

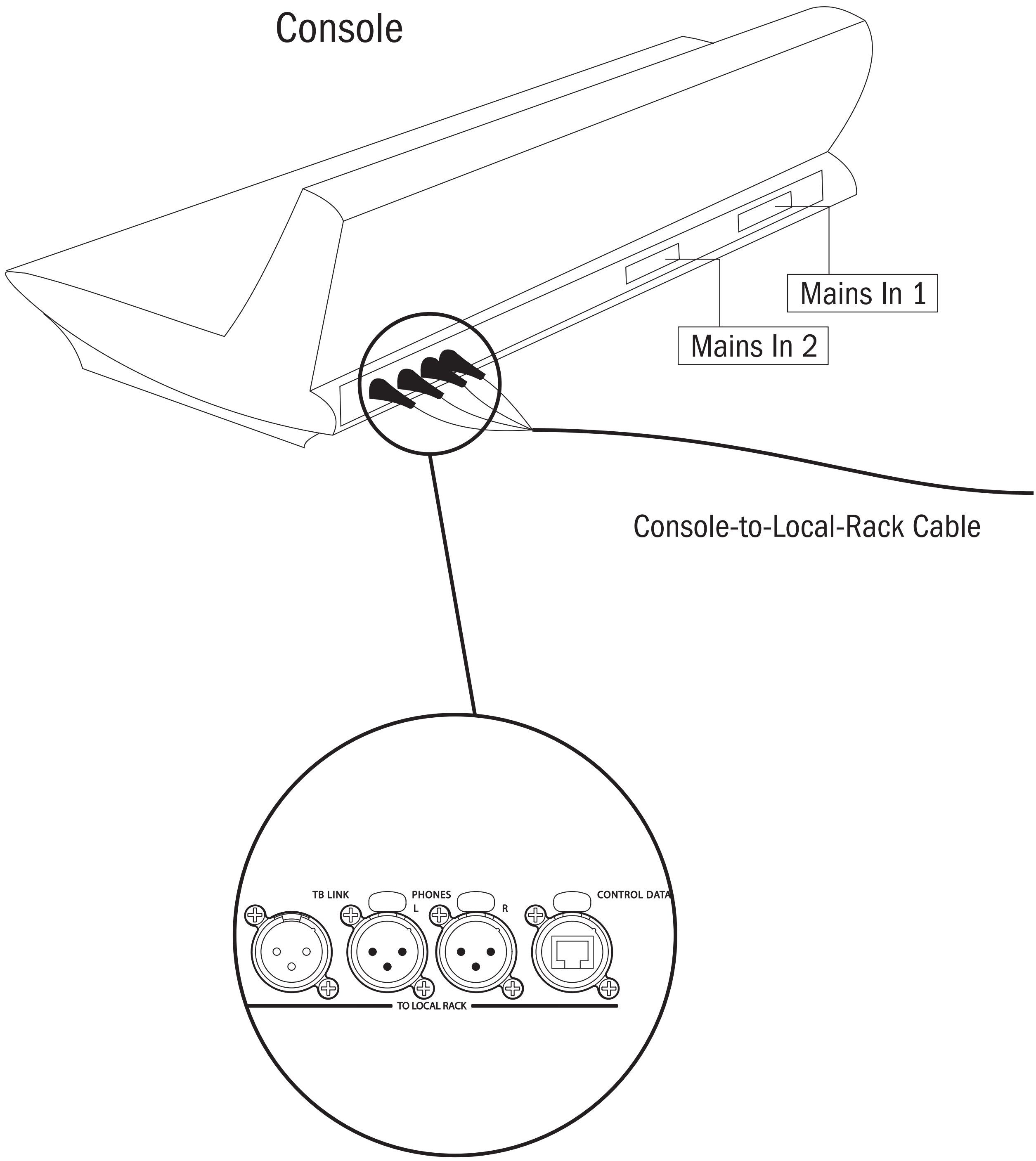
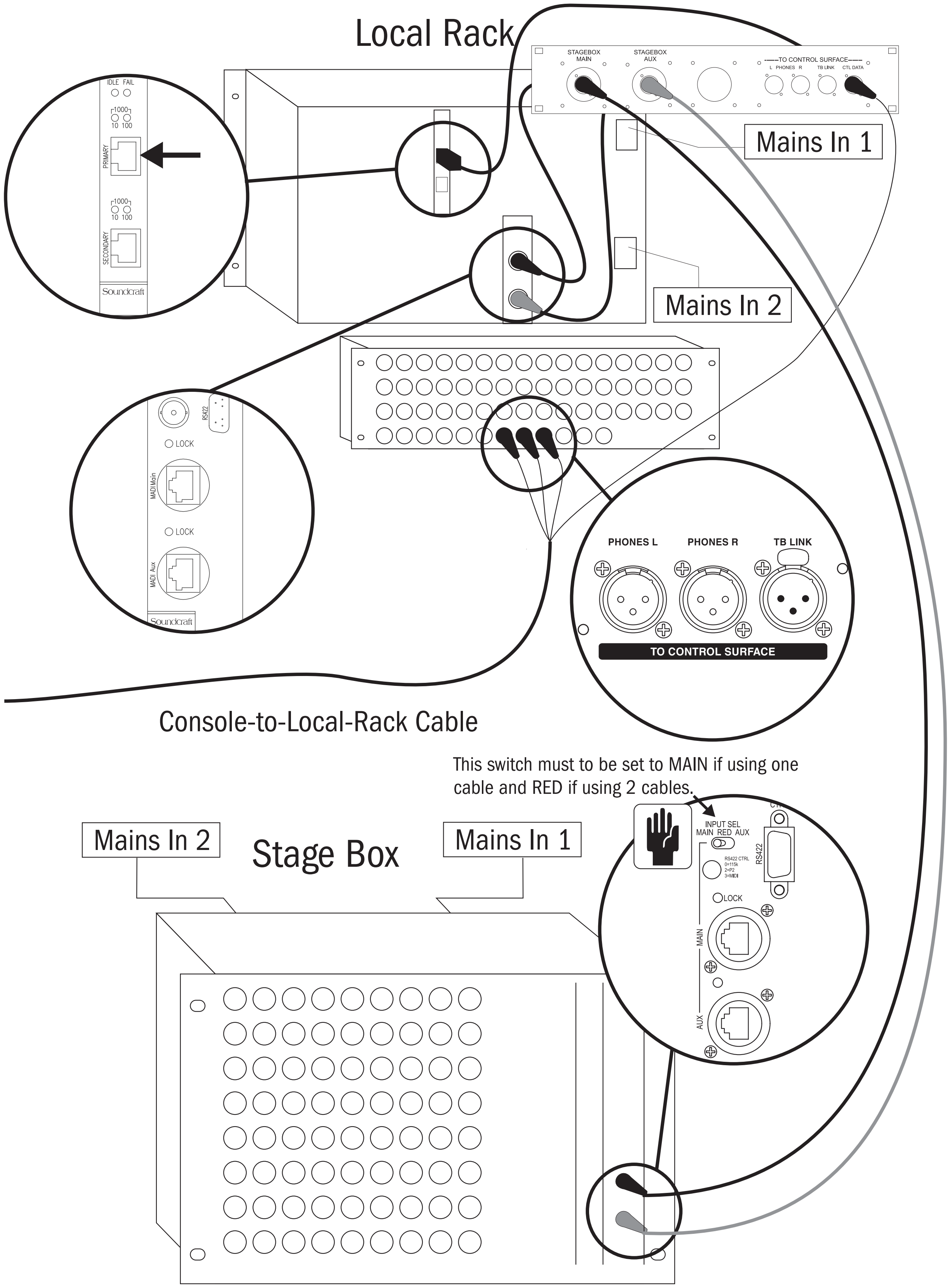


Figure 2-10a: System Interconnections.



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

CLEAN FILTERS REGULARLY!

Figure 2-10b: System Interconnections.

CONVENTIONS USED IN THIS MANUAL

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

GENERAL RULES

* Pressing a [SETUP] key whilst in that SETUP function will exit that function immediately.
* Vistonics™ {EXIT} buttons close the page immediately.
* Parameter changes made by the user are processed immediately.
* Grey-out is used to show that an audio function block is bypassed.
* In order to allow the pre-setting of parameters it is possible to change the parameters and states even if the block is greyed out, e.g. EQ filters can be switched on/off and parameters can be changed even if the Equaliser is switched off with the EQ {IN} key.

SCREENS

Input Screens

The Input Screens are divided into logical areas and fields as shown below.

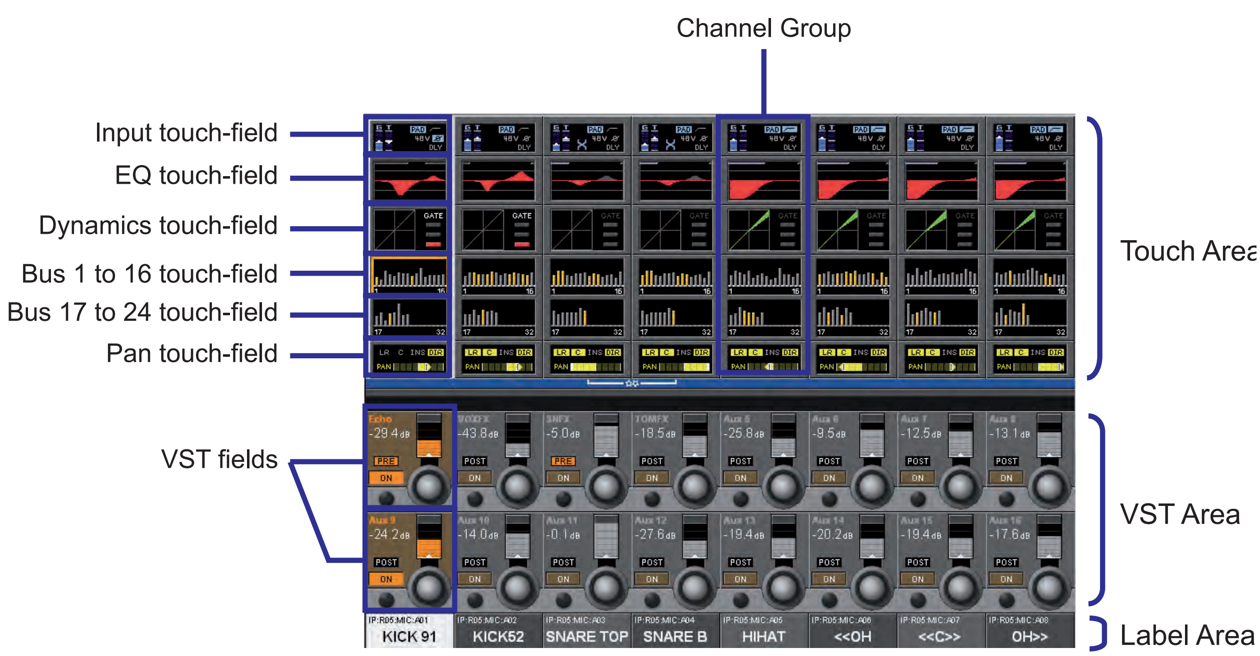


Figure 3-1: Input Bay Screen Areas and Fields.

Further details about these areas and fields can be found in chapter 4.

Note: each TFT screen may have up to 4 permanently bright or permanently dark pixels, and still be within the TFT screen manufacturer's specifications.

Screen Colour Codes

The following table shows the colours used in their corresponding context:

Function	Colour
Audio Processing	
Input Functions	Blue
Equaliser	Red
Filter	Dark 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
Mute Group Patch	Red
Input Patch	Blue
Output Patch	Red
Control	Grey

For the Soundcraft FaderGlow™ colours see chapter 18. Soundcraft FaderGlow™ (Pat. Pend.) is a unique feature that gives the user an additional level of status overview, and can significantly reduce operating errors.

VISTONICS II™ KNOBS

If a function is assigned to the button, the state is indicated like this:

OFF ON

Figure 3-2a: VST Button Status Indication.

If the button can open a configuration page in the Touch area it looks like this:

Normal Page Is Open

Figure 3-2b: VST Buttons Which Can Open A Configuration Page.

AUDIO FUNCTION STATES

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.

Function Active Function Disabled

Figure 3-2c: Active/Disabled Functions.

MOMENTARY/LATCHING CONTROL ACTION

The physical keys on the desk have both a momentary and a latching action. If a key is pressed and released within approximately one half of a second, the control will latch. If the key is held down for longer, and then released, the control will return to its original state as the key is released.

The keys on the Vistonics area also operate in the same way.

The touch-screen buttons/areas operate in a latching mode only.

SOLO/SEL KEYS

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.

LABELLING

General

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

Fig 3-3: The On-screen Keyboard.

Channel Labels

By default the channels are labelled CH-1 to CH-96.
Soundcraft Vi Series™ uses long labels for the Screens and short labels for the LCDs. Long labels can contain up to 10 characters, whereas short labels are restricted to 6 characters.

Figure 3-4: The Channel Label Page.

Changing The Channel Label

* Press the <INPUT> touch field to open the Input Page.
* Enter the channel label page by pressing {CH LABEL} on the Input Page.
* Type the long name (only valid characters are possible).
* Adjust the short name if necessary by touching the <Short Label> on the screen or by using <TAB>.
* Pressing backspace twice while in the short label field will copy the long lable into the short label field, with characters in red indicating that they will not be displayed.
* Leave the page with <ENTER>, or by pressing {CH LABEL} again.

<TAB> toggles the cursor between long and short fields.

BUS CONFIGURATION

The most important configuration of the Soundcraft Vi Series™ is the bus configuration, and this is done using the [ALL BUSSES] view on the four input bays.

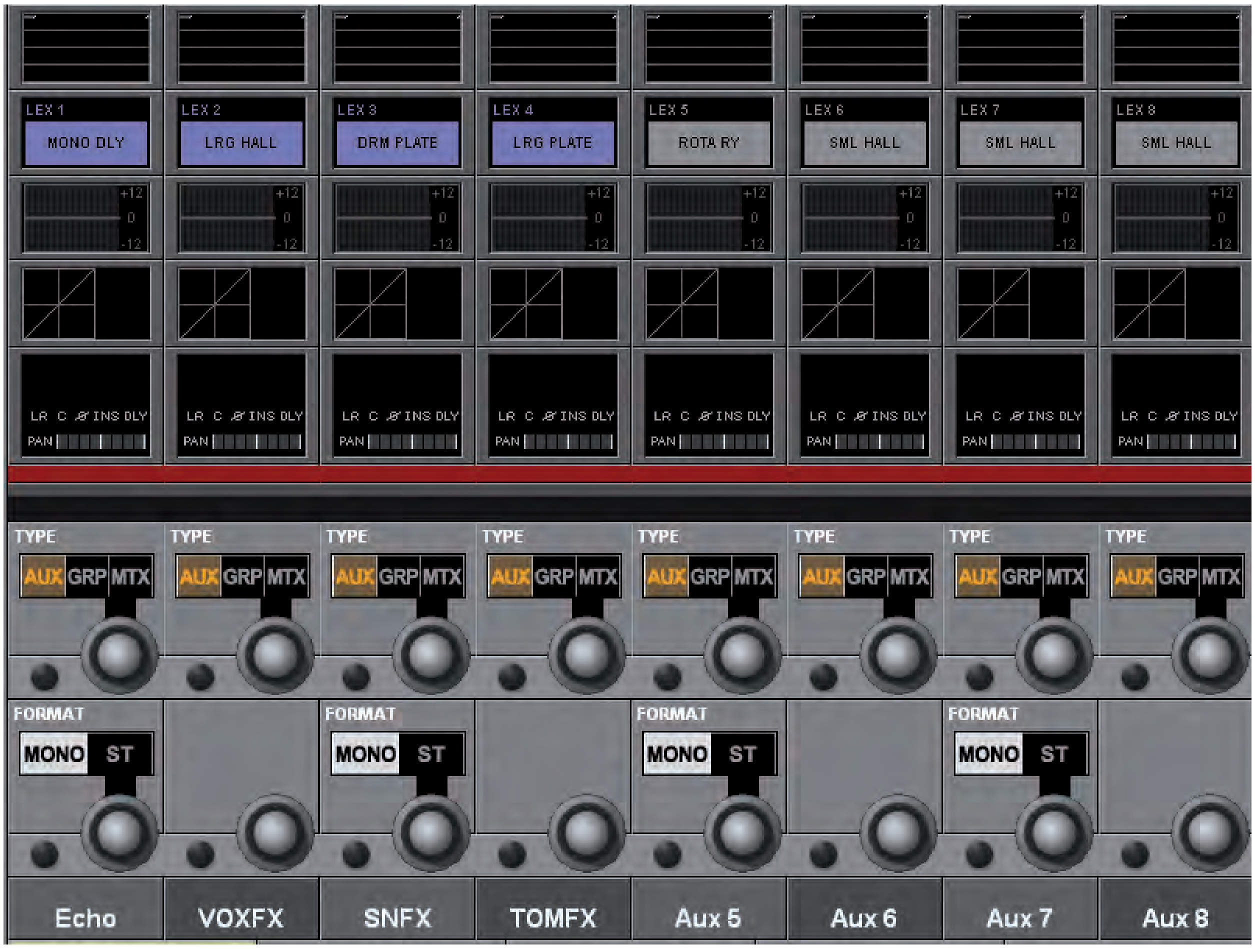


Figure 3-5: The Input Screen If [ALL BUSSES] Layer Is Active.

The TYPE field can be set to one of three values: AUX, GRP or MTX, the encoder is used to change the Bus Type. If Aux is selected and if the format field, see below, is set to stereo then the {CHPAN} field enables the stereo Aux send signal to follow channel pan, rather than have its own pan control.

The FORMAT field (only odd busses) is available for Aux and Grp busses. The field can be set to Mono or Stereo. The Encoder changes the setting of the Audio Format field. If the field is set to stereo the next even numbered bus will not be shown.

In the ALL BUSSES Layer, all 32 Busses are shown from left to right on the four input bays on the Vi6. i.e., Bus 1 is mapped to the left-most strip, while bus 32 is mapped to the right-most strip of the Control Surface (For Vi4, 24 busses will be shown, access other busses via the masters or output meter screen).

Vi Series supports the following Bus Types:

- AUX Mono
- AUX Stereo
- Group Mono
- Group Stereo
- Matrix Mono

GANG

General

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.
For example, if Input Channels are ganged, then a parameter change of a function will be applied to all other ganged channels in an offset manner. For example, adjusting any rotary parameter or fader within a gang will add that offset to, or subtract it from, all other channels in the gang. Pressing a switch will change all other channels whose switches are not currently in the resulting state, to that state. From that point on, further presses will result in all switches changing mode together.

Creating A Gang

* Activate the GANG Mode with [GANG], the [GANG] key will glow blue (see Figure 1-4 for the location of the [GANG] key).
* Add/remove a channel by pressing channel's [SOLO/SEL] key. The [SOLO/SEL] becomes blue if the channel is in the gang.
* ADD/Remove a range by pressing the first and last channel [SOLO/SEL] together.

Switching-Off Gang Mode

Once a Gang has been created, it can be de-activated by switching the [GANG] key off. The [SOLO/SEL] keys will return to normal solo operation. The Gang will be stored however, and can be re-activated for further use at any time. Gang member settings are independent of the console's snapshot automation, but are stored when the console is powered off.

Clearing A Gang

* Press and hold any active (blue) [SOLO/SEL]
or
* Leave the GANG mode with [GANG].

Gang All Input Channels

* Press and release [GANG] to switch Gang mode on, and then press and hold [GANG] until all of the Input [SOLO/SEL] keys turn blue. This selects all channels (including hidden layers) to the gang. When all of the Inputs are Ganged their [SOLO/SEL] keys turn blue.

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.

It is recomended that Gangs are cleared down after use, particularly if GANG ALL is used.

SIGNAL FLOW

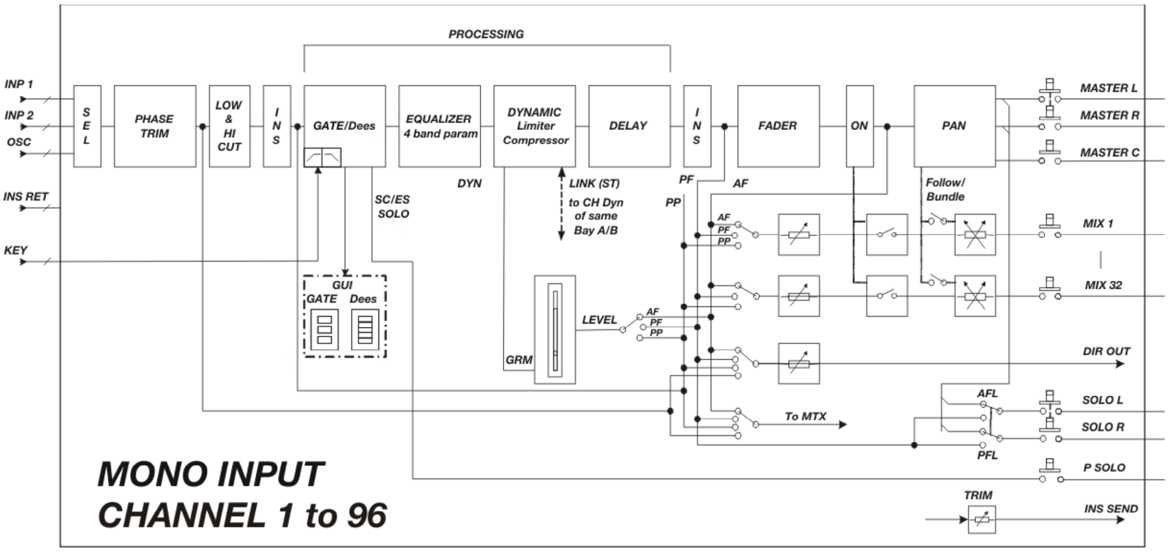


Figure 4-1: The Signal Flow In An Input Channel.

Only one of the two insert points can be used per channel at any time.

INPUT CHANNEL STRIP

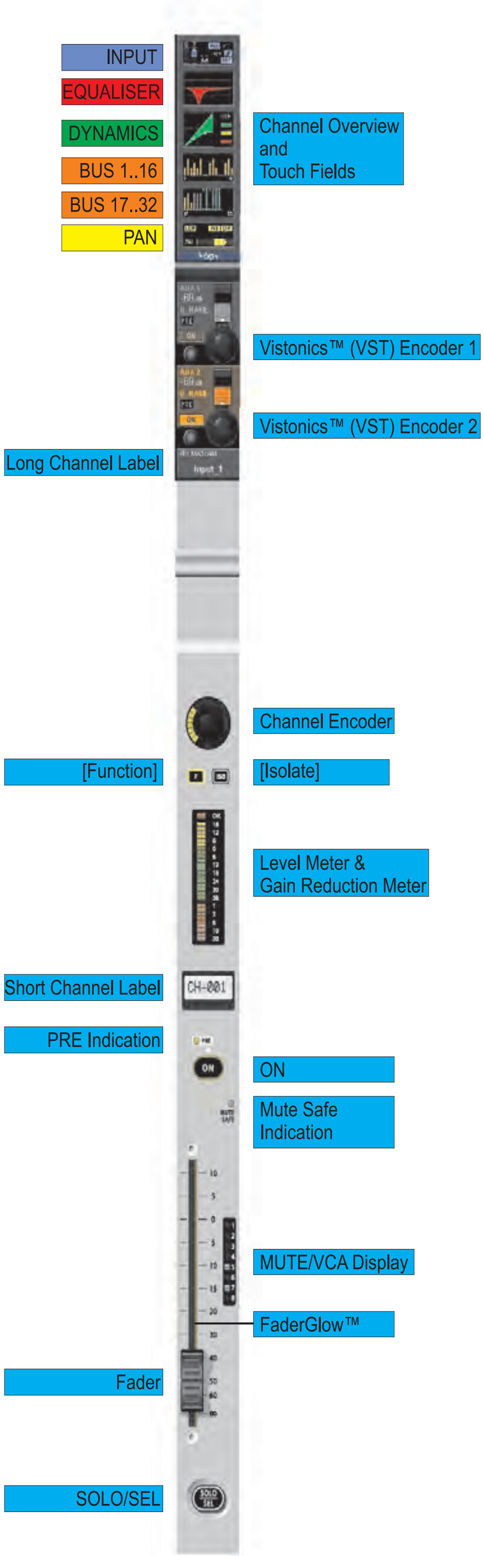


Figure 4-2. Input Channel Strip.

The Level Meter reads from -36dB to +18db, The GRM (Gain Reduction Meter) reads from -1 to -20dB.

INPUT CHANNEL TOUCH FIELDS

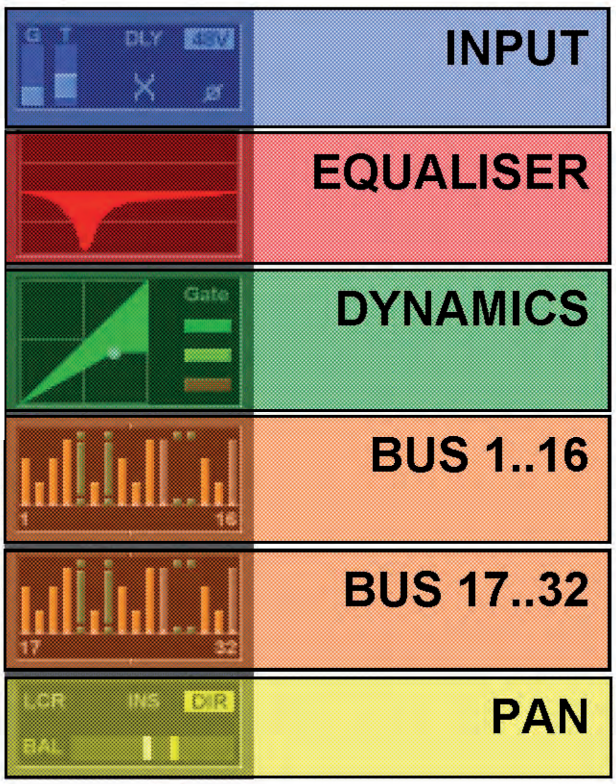


Figure 4-3.

Change A Parameter Of An Input Channel

* Press the desired touch field, the corresponding VST area will open,
* change the parameter.
* Press the touch field again to go back the default VST view OR
* Press another touch field.

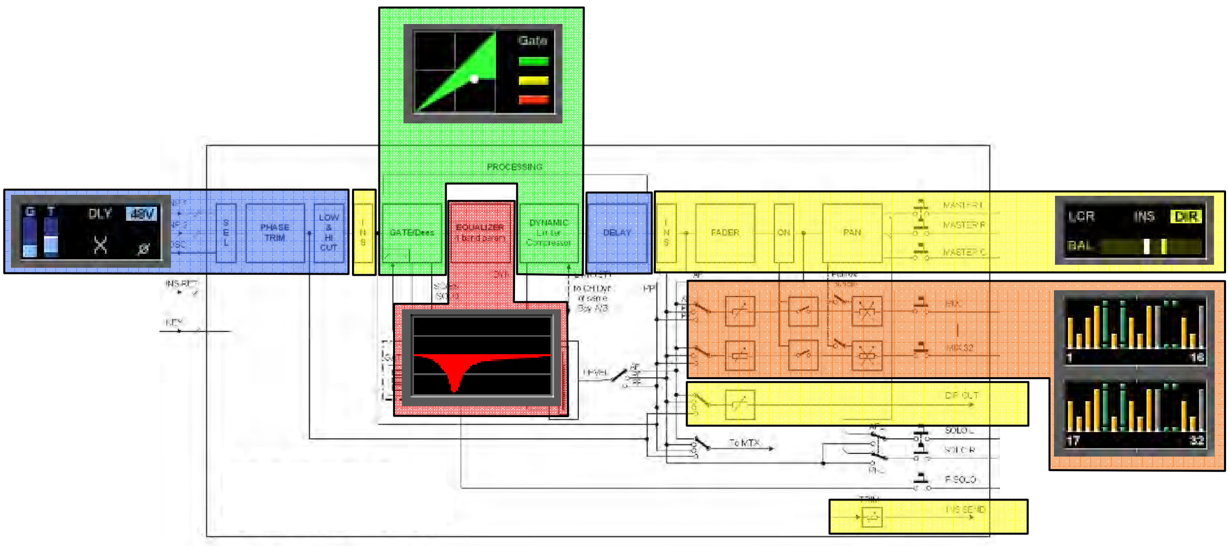


Figure 4-4: the relationship between touch fields and channel functions.

INPUT

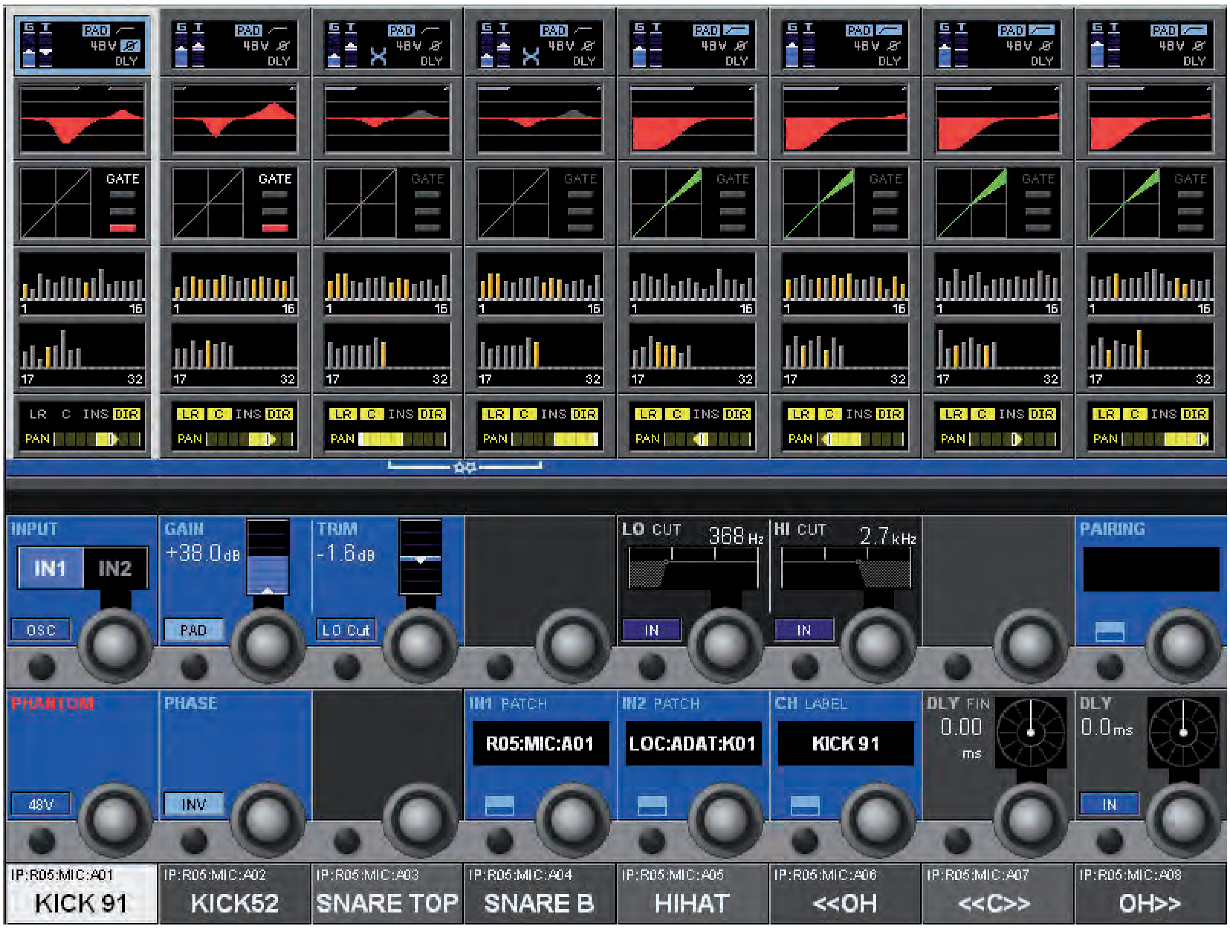
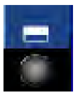


Figure 4-5: The Input Page.

INPUT field

Select source IN1 or IN2. If OSC is active, the central oscillator is patched to this channel and the selector is disabled.

Pressing the {IN1 PATCH} or {IN2 PATCH} VST config key  opens the Input Patch Configuration page.

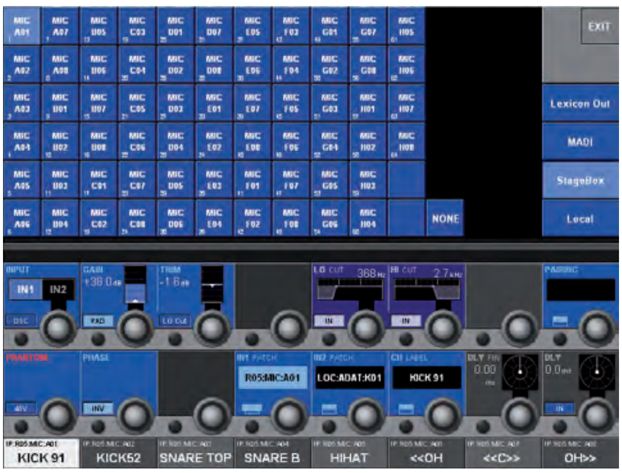


Figure 4-6. Input Patch Configuration Page.

GAIN field
{encoder} adjusts the analogue input gain in the range from +15 dB to +70 dB.
{PAD} reduces the input sensitivity by 20 dB.

TRIM field
Encoder adjusts the digital Input Gain in the range +18/-36 dB
{LO CUT} inserts the pre AD converter analogue low cut filter (only analogue inputs).

LO CUT field
Encoder adjusts the Low cut frequency in the range 20 to 600 Hz.
{IN} switches the Low Cut filter in and out.

HI CUT field
Encoder adjusts the High cut frequency in the range 1k to 20kHz.
{IN} switches the Hi Cut filter in and out.

FORMAT field
If the Channel is paired, the Encoder adjusts the Stereo format, which can be:
LR / RL / LL / RR / MONO.
If the channel is not paired, this field is not displayed.

PAIRING field
If the Channel is paired the label of the paired channel is visible.
{Its VST config button} enters the pairing configuration.

PHANTOM field
{48V} enables the Phantom Power (+ 48 V) for the XLR patched to this input.

PHASE field
{INV} inverts the phase of this channel (180 Degrees).

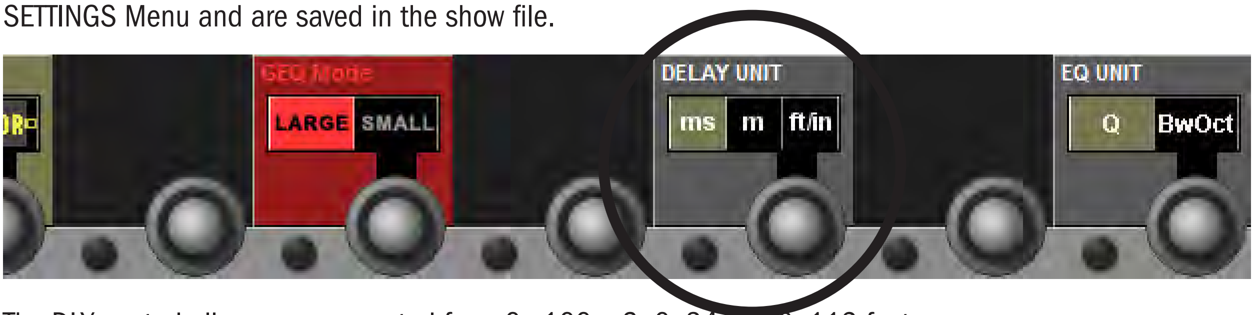
IN1 PATCH field
Displays the source name that is patched to IN1.
{Its VST config button} opens the IN1 patch page (see chapter 10).

IN2 PATCH field
Displays the Source name that is patched to IN2.
{Its VST config button} opens the IN2 patch page (see chapter 10).

CH Label field
Displays the channel label.
{Its VST config button} opens the channel label configuration page.

DLY & DLY FIN field

Encoder changes the input delay in milliseconds, metres, or feet & inches. These units are set in the SETTINGS Menu and are saved in the show file.



The DLY control allows coarse control from 0.. 100 mS; 0..34mts; 0..112 feet
The DLY FIN control allows fine adjustment in 0.02ms/.02 mtrs/0.1" steps.
{IN} enables the delay function.

Distance conversions assume a fixed temperature of 20°C/68°F. This is not adjustable.
Changing the unit controls (ms/ft etc) does not affect any current value and therefore will not affect audio.

STEREO CONFIGURATION

Pairing of input channels

Figure 4-7: The Pairing Page.

The available channels on the two fixed Layers A (upper row) and B (lower row) are displayed on the touch screen.
Pairing candidates are the unused left and right neighbours and the vertical neighbour in the same Bay. Existing Pairings will be shown greyed out. It is not possible to pair with a channel in the next bay.

HINT: Selecting an input channel [SOLO/SEL] moves this page to the desired channel in the corresponding bay.

Pair An input Channel

* Enter the pairing page by pressing {PAIRING}  in the Input Page, all possible pairing candidates will be shown.
* Select the desired pairing candidate, all channel parameters will be copied to the paring candidate, links will be set. The meter overview in Master screen shows the pairing information
* Optionally change the copy direction with <FROM>, <TO>.
* Leave the page with {EXIT} or {PAIRING}.

HINT: If a destination candidate is used in another pair you must first release that pairing.

Stereo Busses

Busses can be configured to work as a Stereo Bus. In this case the Bus Master is represented in one Strip on the surface. The advantage is that more masters can be directly accessed (rather than tying up two Output controls).
In the Encoder section stereo bus masters also use only 1 VST Field.

EQUALISER

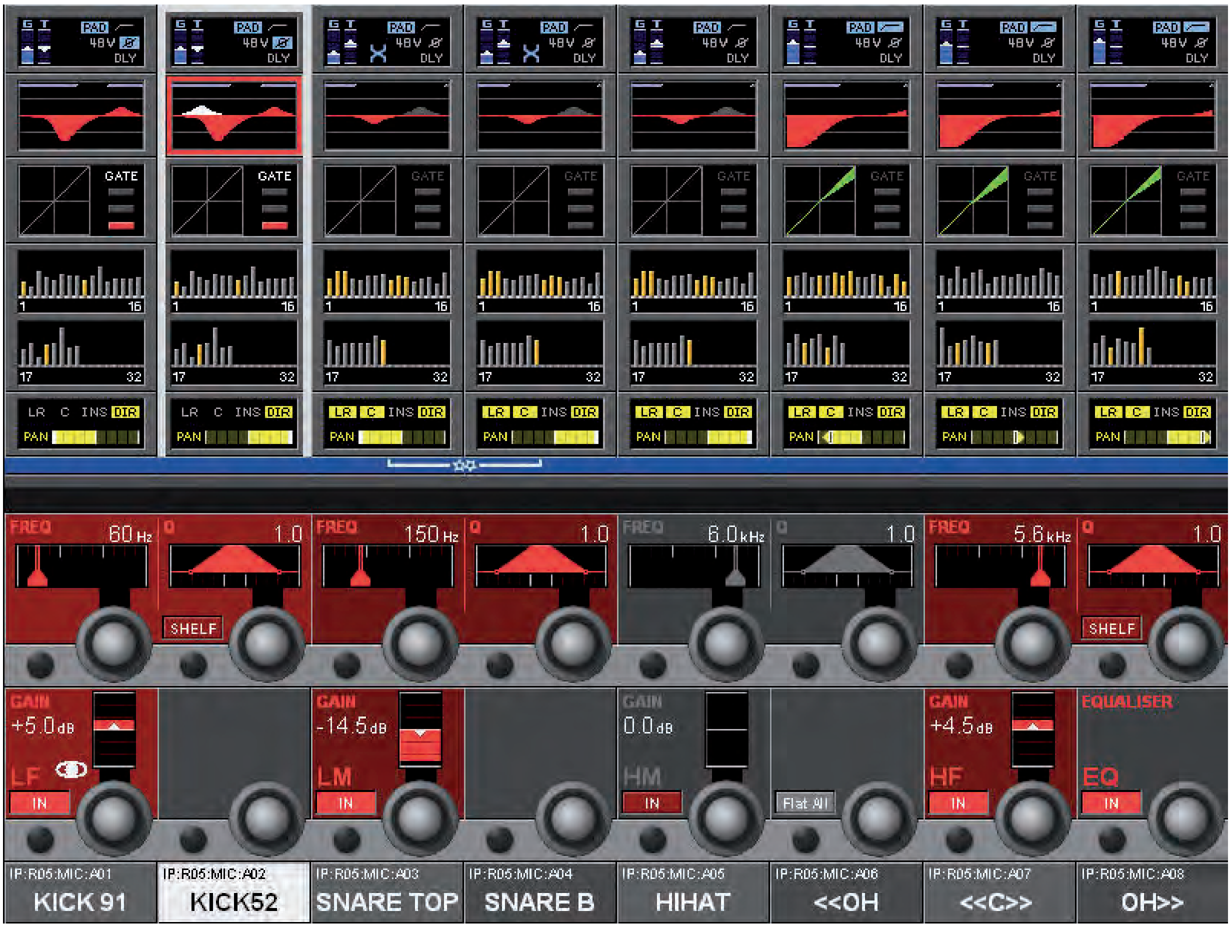


Figure 4-8: The Equaliser Page.

General

The Equaliser contains 4 full parametric Bands. All four parametric Equaliser Bands operates over the full frequency range. The characteristics of the LF and HF bands can also be set to SHELF mode.
The red Equaliser Graph in the Equaliser touch field represents the overall frequency curve. It contains:

* Low Cut and High Cut Filters
* Equaliser with the 4 bands

Additionally, two blue bars at the top of the channel strip Equaliser touch field clearly indicate the Low and High Cut filter frequencies. (The filters are adjusted from within the INPUT Vistonics™ II page.)

Equaliser Band Highlight

If you touch one or more parameter encoders, the corresponding Equaliser band(s) will be indicated by the red overall graph being overwritten with a white graph that represents only the touched band(s). This is useful for identifying which part of a curve is associated with which EQ bands. The representation will return to the default when you leave the page.

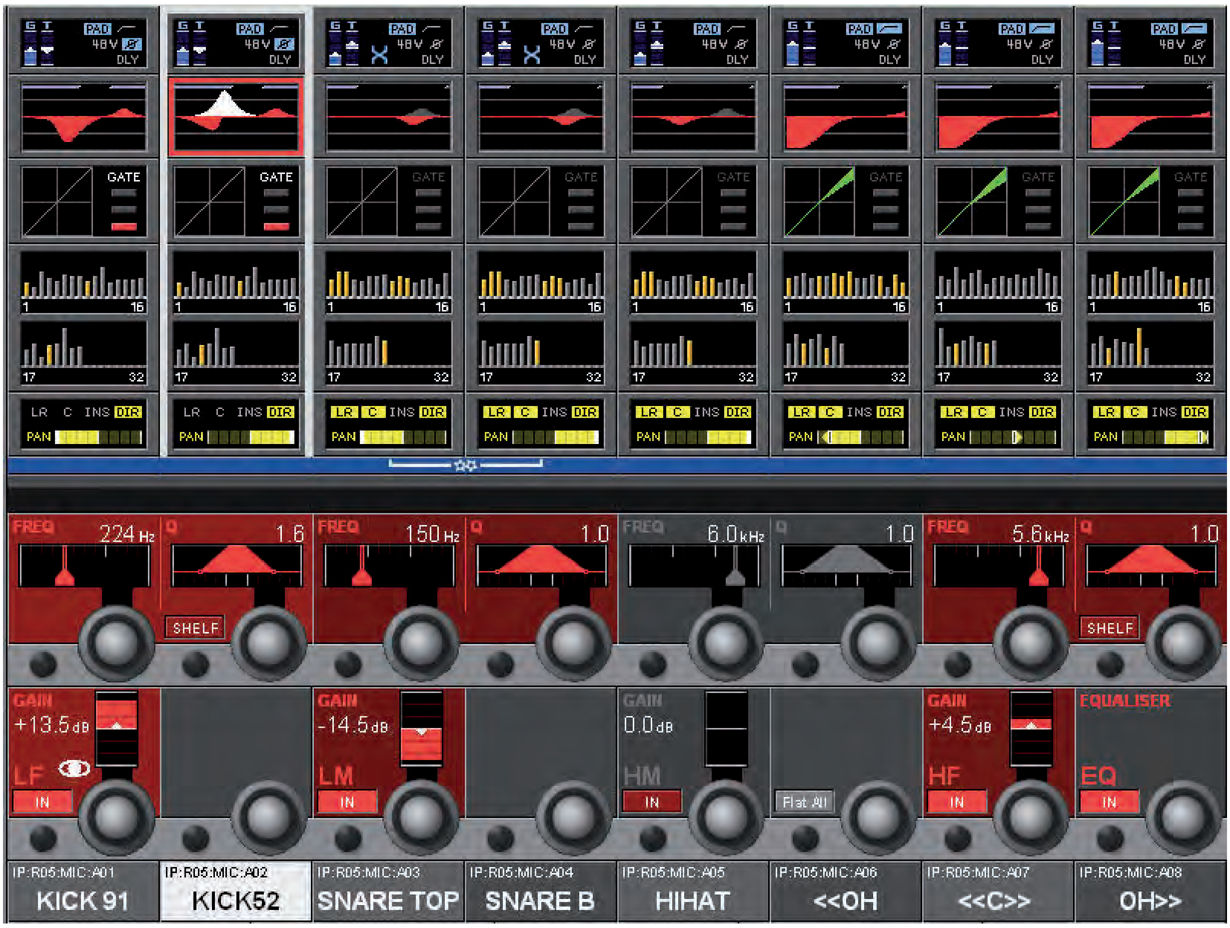


Figure 4-9: The Equaliser Page After Touching An LF Encoder.

BAND Field

All four bands contain a GAIN, FREQ, Q encoder and an IN switch.

GAIN

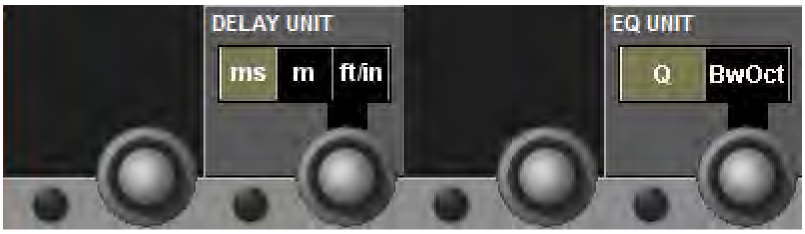
Encoder adjusts the Gain in the range +/- 18 dB. {IN} enables the Equaliser band.

FREQ

Encoder adjusts the frequency in the range 20 Hz .. 20 kHz.

Q

This control may be set to operate as Q or BANDWIDTH in the SETTINGS menu. These settings are saved in the show file.



IN Switch

The channel Encoder adjusts the Q (Width) in the range 0.7 .. 15, or bandwidth from 0.2 to 4 octaves.
Q may only be adjused in BELL Mode.

Changing the unit controls (ms/ft etc) does not affect any current value and therefore will not affect audio.

{IN} enables this Equaliser band.

SHELF Switch

{SHELF} changes the response characteristic to SHELF. (12dB/oct)

EQUALISER Field

{IN} enables the full Equaliser section (without filters). For preparation, all Parameters including the band IN switches can be set, even if the Equaliser is switched off.

DYNAMICS

Figure 4-10: The Dynamic Page In GATE Mode.

General

The Dynamics section contains a GATE with key filter a Compressor and a Limiter. Alternatively the Gate can be set to operate in De-esser mode.

GATE Function

The Gate contains a side-chain input with filters, a Key listen SOLO and it can also operate in ducking mode.

State Indication

The Gate status is indicated on the right side of the Dynamics touch field:

* ON green
* HOLD yellow
* OFF red

THRS Field
Encoder adjusts the threshold value between -40 db to +18db.
{IN} enables the GATE.

ATCK field
Encoder adjusts the attack time in the range 10µS to 957mS.
{DUCK] sets the GATE to inverse mode (ducking mode). (Was {INV} in software prior to V4.7).
Engaging this button inverts the gain control function of the Gate and allows an external sidechain signal to be used to reduce the level of the main channel signal, by the amount set using the RNG control, and with the time constants set with the ATK, HOLD and REL controls.

HLD field
Encoder adjusts the hold time in the range 2.2mS to 2S.

REL field
Encoder adjusts the release time in the range 2.2mS to 3.7S.

RNG field
Encoder adjusts the attenuation RNG value in the range 0 to -60dB.
{SC SOLO} switches the side-chain signal to the solo bus.

LO CUT
Encoder adjusts the frequency of the side-chain signal Lo Cut filter.
{IN} enables the Low cut filter.

HI CUT
Encoder adjusts the frequency of the side-chain signal Hi Cut filter.
{EXT Key} enables an external key signal, which is selected via the {KEY} key associated with the MODE field below; otherwise the internal signal is used for triggering the gate.

MODE field
Encoder switches the operational mode between GATE or Deesser.
{Key} opens the key signal patch page -see chapter 10.

DE-ESS Function

A de-esser is normally used to reduce the sibilance (“s” components) in a singer’s voice.
Soundcraft Vi Series™ includes a real de-esser function that works as a dynamically controlled filter. The filter can be set using the FREQ and Q encoders.
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.

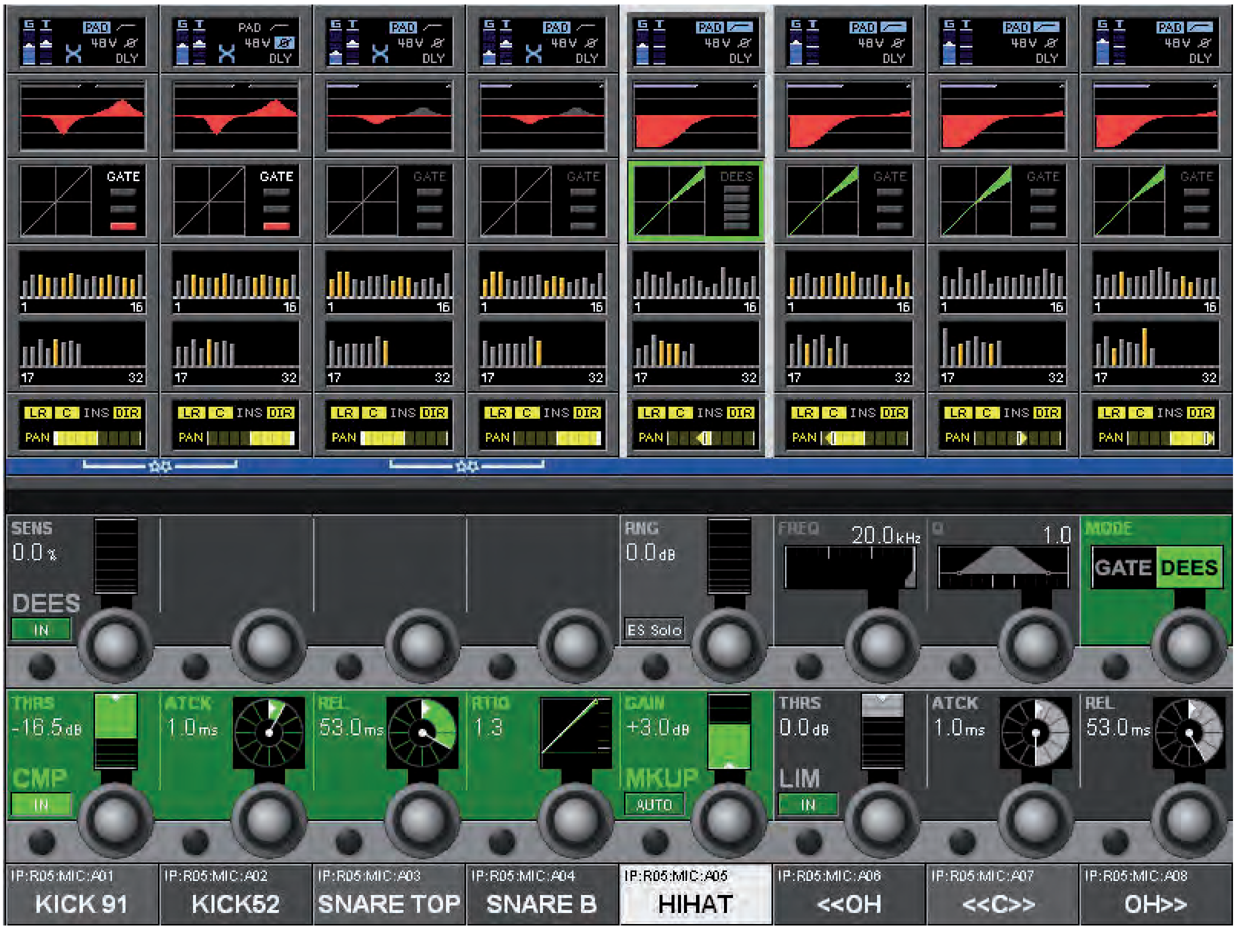


Figure 4-11: The Dynamic Page In DE-ESS Mode.

Gain Reduction Meter

The De-esser GRM, a 5-segment bar-graph is located on the right-hand side of the dynamic touch field on the screen.

SENS Field

Encoder adjusts the effect’s sensitivity value between 0 – 100%.
{IN} enables the DE-ESSER.

ES SOLO Field

{ES SOLO} switches the filtered processing signal to the solo bus.

FREQ Field

Encoder adjusts the centre frequency of the dynamic de-esser filter.

Q Field

Encoder adjusts the width of the dynamic de-esser filter.

MODE

Encoder selects the operational mode, either GATE or De-esser.

COMPRESSOR Function

THRS Field

Encoder adjusts the threshold value between -40 db to +18dB.
{IN} enables the COMPRESSOR.

ATCK Field

Encoder adjusts the attack time in the range 0.5mS to 98.6mS.

REL field

Encoder adjusts the release time in the range 5.5mS to 5S.

Ratio field

Encoder adjusts the ratio in the range 1:1 to 20:1.

MKUP

Makeup adjusts the overall output level from the Limiter and Compressor sections.

GAIN

Encoder manually adjusts the output level to compensate for gain reduction.
{AUTO} automatically adjusts the output level depending on the settingd of the THRS and RATIO controls.

LIMITER Function

THRS Field

Encoder adjusts the threshold value between -40 db to +18dB.
{IN} enables the LIMITER.

ATCK Field

Encoder adjusts the attack time in the range 10µS to 98.6mS.

REL Field

Encoder adjusts the release time in the range 5.5mS to 957mS.

BUS

Figure 4-12: The Bus Page.

General

The first Bus page contains the controls for busses 1-16, while the second page contains the controls for busses 17-32.
What these fields look like depends on the Bus configuration.

AUX (Mono)

Encoder adjusts the send level to this BUS. Pre indicates the PRE/post state. {ON} enables the send.

AUX (Stereo)

Left Encoder adjusts the send level to both busses, while the right encoder adjusts the pan to these busses.
HINT: If the channel is paired, the right encoder adjusts the Balance to the Busses. If the 'follow channel pan' option was activated in the bus configuration, there will be no function on the right encoder.
PRE indicates the pre/post state. {ON} enables the send.

GRP (Mono)

{ON} routes the signal to this bus.

GRP (Stereo)

{ON} route the signals to both busses.

Empty

If a bus has been configured as a Matrix, it is not visible in this page.

PANNING

Figure 4-13: The Panning Page In LR Mode.

General

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

PAN Function LR Mode

PAN Field

Encoder sets the channel pan to the masters. If the channel is paired, the balance can be adjusted. See Audio Format / Pan / Panning, on pages 4-17 & 4-18.

MASTER LR

{ON} routes the channel signal to the Left and Right master busses.

MASTER C

{ON} routes the channel signal to the Centre master bus.

MODE

Switches the pan mode between LR or LCR. This field is not available for stereo-paired inputs.

PAN Function LCR Mode

If the Pan MODE is set to LCR an additional WIDTH field is displayed. (If the channel is paired, it is not possible to set the PAN mode to LCR.)

Figure 4-14: The Panning Section In LCR Mode.

PAN Field
Displays the pan setting. Encoder sets the channel pan to the three masters. See Audio Format / Pan / Panning on pages 4-17 & 4-18.

WIDTH Field
In LCR mode, the encoder adjusts the level of an additional amount of signal sent to both left and right outputs.

MASTER LCR
{ON} routes the channel signal to the Left, Right and Centre masters.

AUDIO FORMAT

General

The Soundcraft Vi Series™ can handle three types of audio format:

* MONO
* STEREO
* LCR

Input Channels

Soundcraft Vi Series contains up to 96 MONO Input channels. A STEREO Input can be built by horizontally or vertically pairing two input channels in the same bay.
Vertically and horizontally pairing can be used at the same time. See Pairing of input channels.

Mix Busses

The 32 MONO Busses can be configured to work as Mono or as odd/even paired Stereo Busses.

Masters

The three Masters L, R, C can be used as LCR Masters if the Pan mode of the input channels is set to LCR mode, otherwise L and R works as stereo output and the C can be used as an independent Mono Master.

PAN/BAL

The following table shows the destination level in relation to the PAN/BAL settings

MODE	Left position		Middle position		Right position	
	Left	Right	Left	Right	Left	Right
PAN OFF	- 3 dB	- 3 dB	- 3 dB	- 3 dB	- 3 dB	- 3 dB
PAN ON	0 dB	- ∝	- 3 dB	- 3 dB	- ∝	0 dB
BAL OFF	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
BAL ON	+ 3 dB	- ∝	0 dB	0 dB	- ∝	+ 3dB

HINT: If the PAN or BAL function is switched off, the gain is the same as if you had set the Encoder to the middle position.
If all TRIM, Faders and so on are in the 0dB position the outputs from the L and the R Masters are 3 dB lower than a MONO Input signal.

MONO PAN

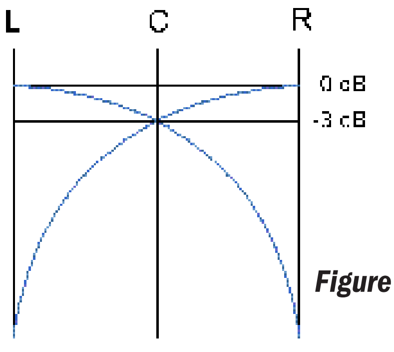


Figure 4-15a.

STEREO BAL

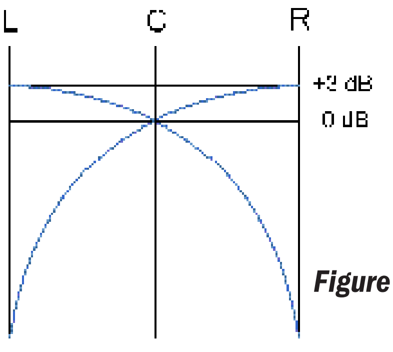


Figure 4-15b.

If a panning mode is set to LCR, then the WIDTH function will become active.

LCR PAN with WIDTH = 0

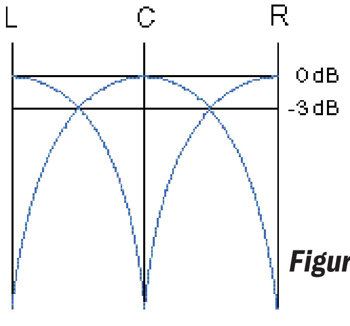


Figure 4-15c.

LCR PAN with WIDTH = 100

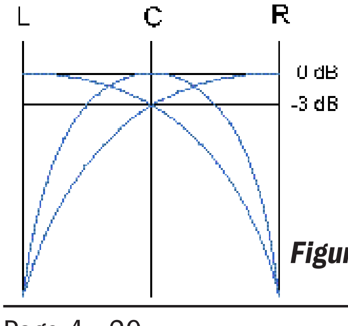


Figure 4-15d.

INSERT Function

Refer to Figure 4-13.

INSERT Field

Displays the label of the selected insert from the pool.
{Its VST config button}opens the Insert Pool select page.
Refer to page 10-6 for details about setting-up the Insert Pool.

POINT Field

Displays the actual point where the Insert is placed in the channel.
Encoder changes the point between:

* Pre Processing (EQ&DYN)
* Pre Fader.

TRIM Field

Encoder trims the insert send level in the range +- 18 dB.
{IN} activates the Insert.

Direct Out Function

GAIN field

Encoder sets the Direct Output send level.
{ON} activates the Direct Output.

POINT Field

Displays the actual point in the channel's signal path from where the Direct Output is taken.
Encoder selects the point between:

* Pre Filter
* Pre Processing
* Pre Fader
* Post Fader.

PATCH Field

Displays the patched Output.
Its {VST config button} opens the Direct Output patch configuration page (see page 10-8).

ACCESSING CHANNELS

Channels may be accessed using the faderstrips, and also by touching the meter screens on the master screen.

Soundcraft Vi6: Access 96 inputs using the 3 fixed (or user) layers, each accessing 32 channels.

Soundcraft Vi4: You can now access inputs 1-24, 33-56 and 65-88 using the 3 fixed layers.

Method 1: Use the Input Meter display on the touchscreen in the Control Bay to select the additional channels in groups of 8, and assign them temporarily to the right-hand 8 input faders. Note that the input channels that cannot be seen on the fixed layers A/B/C are indicated by a white box around these input meters. This method is good for quickly accessing any of the 96 channels in the default channel order (25-32, 57-64, 89-96), but has the disadvantage that changing to a different main Input Fader page will cancel these temporarily assigned channel assignments.

Method 2: Assign the desired channels to a User Fader page.

Using this method, the new channels can be placed anywhere within any of the User Input layers, and Method 1 can still be used in combination with this as an additional way of grabbing blocks of 8 channels.

Soundcraft Vi2:

Method 1: Press the Input Fader Page buttons D on the left of the fader panel. This gives 6 pages of faders, the first 3 (A,B,C) are fixed as Channels 1-9, 9-16 and 17-24 respectively, whilst the second 3 (USER 1, USER 2, USER 3) default to Channels 25-32, 33-40, 41-48 but can be customised with the associated Fader Page Setup button.

Method 2: Touch the Input Meter screen on the block of 8 meters corresponding to the channels required. This method temporarily takes priority over the setting made using the Input Fader Page buttons, and the last selected Input Fader page button flashes to indicate that it has been temporarily replaced by the meter selection.

SIGNAL FLOW

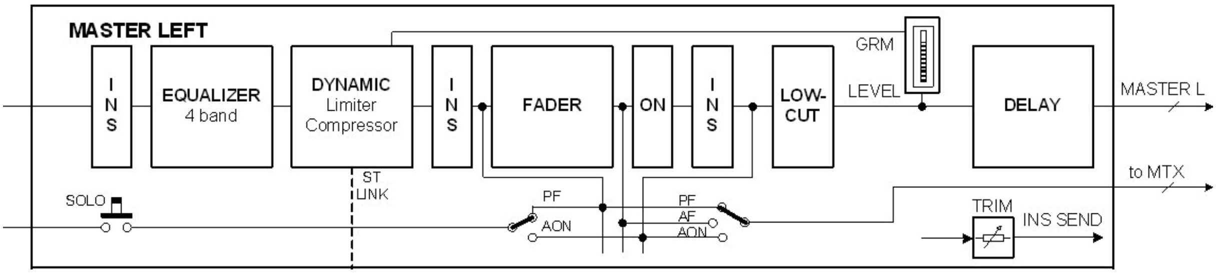


Figure 5-1: The Signal Flow In An LRC Master Or A BUS Master.

Only one of the three possible insert points can be used per master at any time.

GENERAL

The LR and C Masters are always operated with their dedicated master fader strips in the master bay. The parameters of these busses are controlled via the Master Processing Page (see Figure 5-2).

The Soundcraft Vi Series™ allows three different ways to access, control the level of , and change the parameters of, the other 32 output busses. These are:

via the Master Bay's Output Strips, and selecting the fader page* required,

via all of the Input Strips, using the [ALL BUSSES] key,

via the Vistonics™ buttons and encoders on the master section screen.

* For detailed information about Layering see chapter 8.

L,R & C Master Processing

Figure 5-2A: Master Screen Displaying The Master Busses Processing Page.

This page is accessed by pressing the [SEL] key which is located below the L, R & C master faders. The Master processing fields are shown in the Meter Area of the Master Screen.

In order to change the parameters for EQ, Dynamics or Pan, the appropriate area on the touch-screen must be pressed; doing so will open a page whose VST area is similar to the lower half of Figures 5-9, 5-10 or 5-11.

Note: When selecting the PAN area, the page which will be displayed will be similar to Figure 5-11 except that the PAN, MASTER LR and MASTER C fields will not be present.

Note: The output levels of the L R & C Master outputs are always controlled by their dedicated faders.

MASTER EQ LINKING (V2.0 Software and above)

Figure 5-2B : Master Screen Displaying The Master Busses Processing Page with EQ Linking.

The LRC Master busses can have their Parametric and/or Graphic Equaliser sections linked for easier adjustment. Left and Right busses can be linked, or the Centre bus can be added to the linked L&R so that all three busses can be adjusted together. It is not possible to link Left and Centre or Right and Centre.
The linked state is indicated at all times by a pair of white 'gear wheel' icons between the L and R Parametric and Graphic EQ touch fields.
A similar icon with 3 'gear wheels' indicates that the C bus is also linked.
To link or unlink the EQ or GEQ sections

- Press [SEL] below the LRC Master faders to open the Masters strip display.
- Press the {LINK SETUP} button in the bottom right corner of the Master strip display.
- Touch either of the L or R {EQ} touch fields to toggle the linked state on and off for the EQ. Touch the C {EQ} field to add/subtract the C bus EQ to the linked L/R pair.
- Touch either of the L or R {GEQ} touch fields to toggle the linked state on and off for the GEQ. Touch the C {GEQ} field to add/subtract the C bus GEQ to the linked L/R pair.

The fields that are available to be toggled in and out of the linked state are shown with a highlighted white border around the touch field when {LINK SETUP} is active.
Note that the FX fields and the Dynamics fields of the L and R busses are permanently linked, and cannot be toggled in the Link Setup mode.

Default settings

The settings of the EQ and GEQ Linking for the Master busses is stored in the current Show.
The links are set to ON for L,R and C busses for both EQ and GEQ, in the factory default Shows that are supplied with the console.

MASTER BAY OUTPUT STRIPS

The first way of controlling and changing the parameters of the 32 output busses described earlier is as follows.

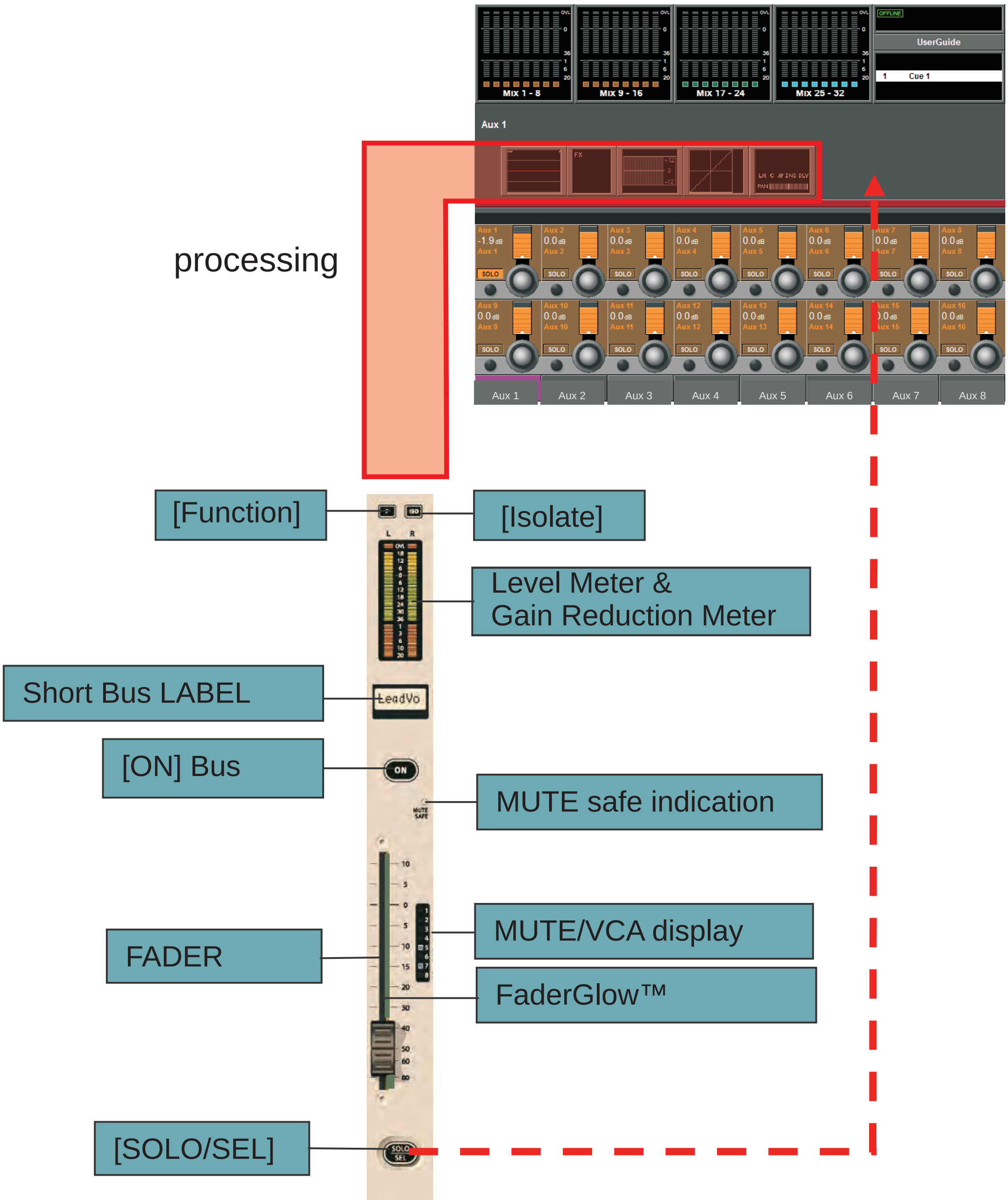


Figure 5-3. A Master Bay Strip.

In order to select the desired output from the 32 possible choices, the correct fader page, A-D, must first be assigned to the master bay (see Figure 8-5 for the keys). Once this is done, the user can select the desired fader to control the output level, and pressing its [SOLO/SEL] key opens the processing Area in the Master screen (see Figure 5-4).

HINT: [METER LOCK] must be off, otherwise the processing strip will not be displayed. If multiple Output Solo are activated the processing for the last-pressed Master Solo is displayed.

Bus Master Processing

Figure 5-4: Master Screen Displaying Bus Processing Page.

The Bus Master processing is shown in the Input Meter Area of the Master Screen. In order to change the parameters for EQ, Dynamics or Pan, the appropriate area on the touch-screen must be pressed; doing so will open a page whose VST area is similar to the lower half of Figures 5-9, 5-10 or 5-11.

HINT: Stereo Busses are linked. Therefore the processing strip will control both channels.

INPUT BAY STRIP USING [ALL BUSSES]

The second method of controlling, and changing the parameters of, the 32 output busses described earlier is as follows.

If the [ALL BUSSES] key is active (see Figure 8-6), the input strips on all of the input bays will be switched to control the 32 output busses.
Once this is done, the user can select the desired fader to control the required output level. In addition, the VST encoders can be used to change the bus type (Aux, Grp or Mtx) and format (Mono or Stereo).

In order to change the parameters for EQ, Dynamics or Pan, the appropriate area on the touch-screen above the required strip must be pressed; doing so will open a page similar to Figures 5-9, 5-10 or 5-11.

In the case of Vi4 and Vi2, not all of the busses will be visible at the same time in the All Busses layer, due to the limited number of faders. In this case, the Output Meter screen can be touched to bring the selected 8 busses onto the far right-hand fader bay.

(In fact this method of selecting busses onto the right-hand fader bay also works in Input fader pages, not only for All Busses).

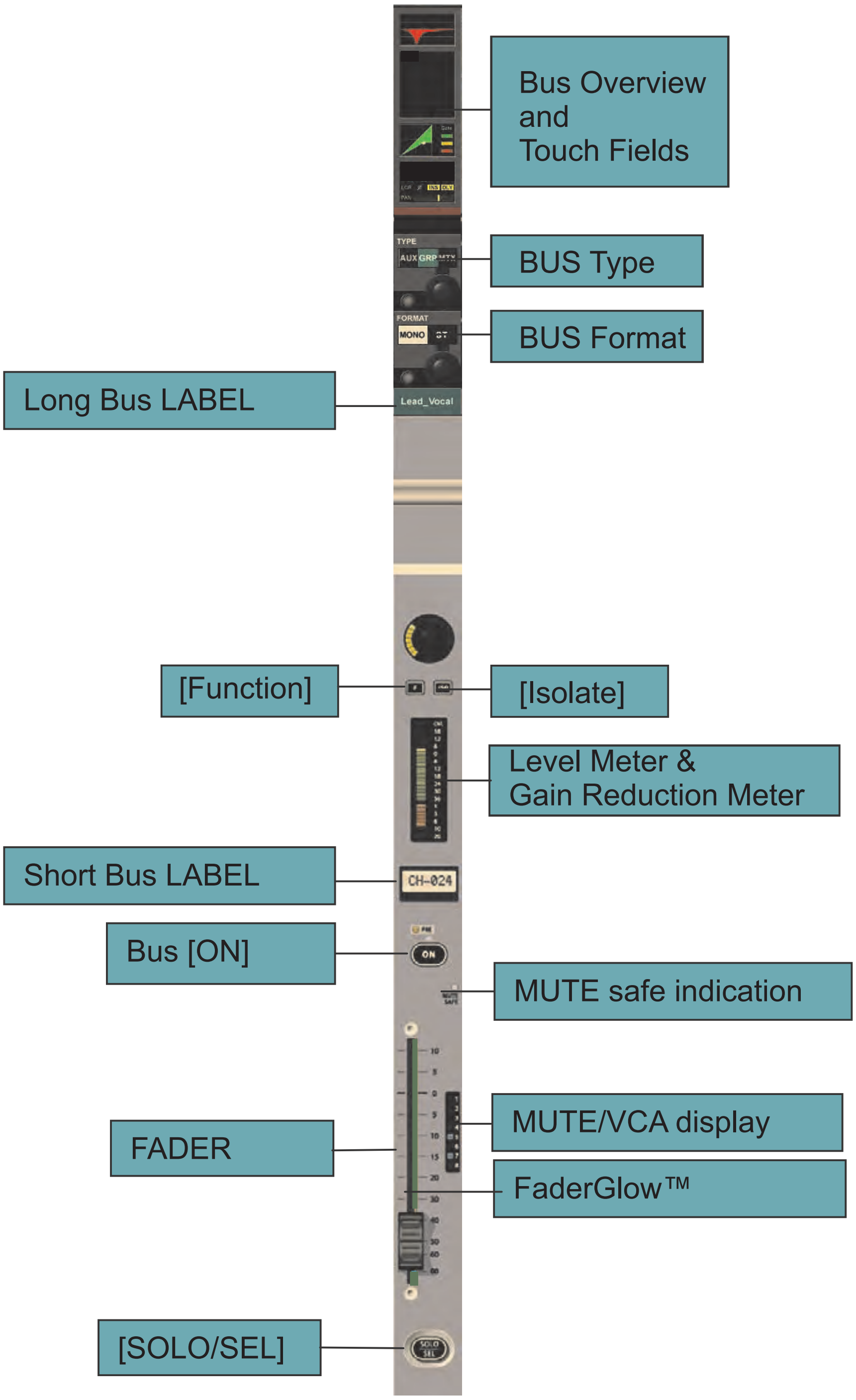


Figure 5-5.Input Bay Strip With [ALL BUSSES] Active.

MASTER BAY VISTONICS™ ENCODERS & KEYS

The third way of controlling, and changing the parameters of, the 32 output busses described earlier is as follows.

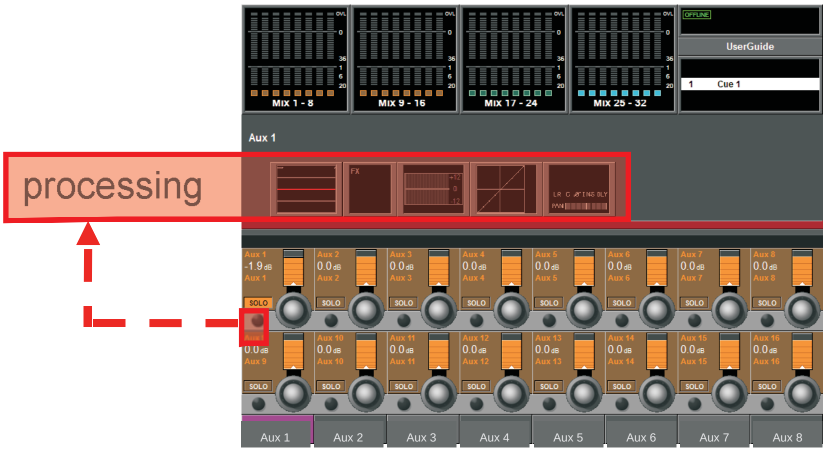


Figure 5-6. Output VST Switches.

The VST encoders control the levels of the displayed output busses. To select the desired range of busses the user must press [Page A] or [Page B] to the right of the screen (see Figure 5-7). The [PAGE A] key displays busses 1-16 in the VST area, [PAGE B] displays busses 17-32.

To change output bus parameters the user must first ensure that the [SOLO SEL] key on the VISTONICS SWITCH FUNCTION panel is active(see Figure 5-7). When this is done pressing the {SOLO} VST key opens the processing Area in the Master VST screen (see Figure 5-6). In order to change the parameters for EQ, Dynamics or Pan, the appropriate area on the touch-screen must be pressed; doing so will open a page whose VST area is similar to the lower half of Figures 5-9, 5-10 or 5-11.

HINT: [METER LOCK] must be disabled, otherwise the processing strip will not be displayed.

VST Key Function

The functionality of the VST key can be set, via the VISTONICS SWITCH FUNCTION panel,to TB Assign, ON/OFF or SOLO/SEL, where SOLO/SEL is the default setting. The function is the same for ALL Encoders in both pages. The [PAGE A] key displays busses 1-16 in the VST area, [PAGE B] displays busses 17-32.

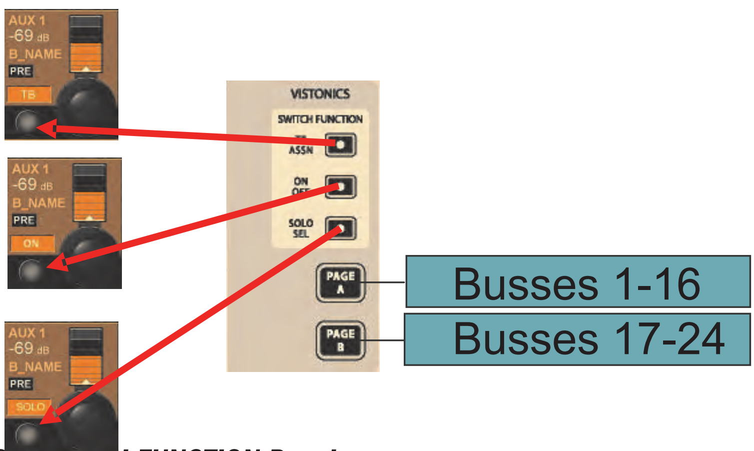


Figure 5-7: VISTONICS SWITCH FUNCTION Panel.

CHANGING OUTPUT BUS PARAMETERS

The following pages shows bus master processing using the Input strip (All Busses layer) mode as an example. Parameter changing is done in the same way if either of the other two ways of accessing bus masters is used.

Changing A Parameter Of A Bus

* Press the desired touch field, the corresponding VST Area will open
* change the parameter
* Press the touch field again to go back to the default VST view
  or
* Press another touch field.

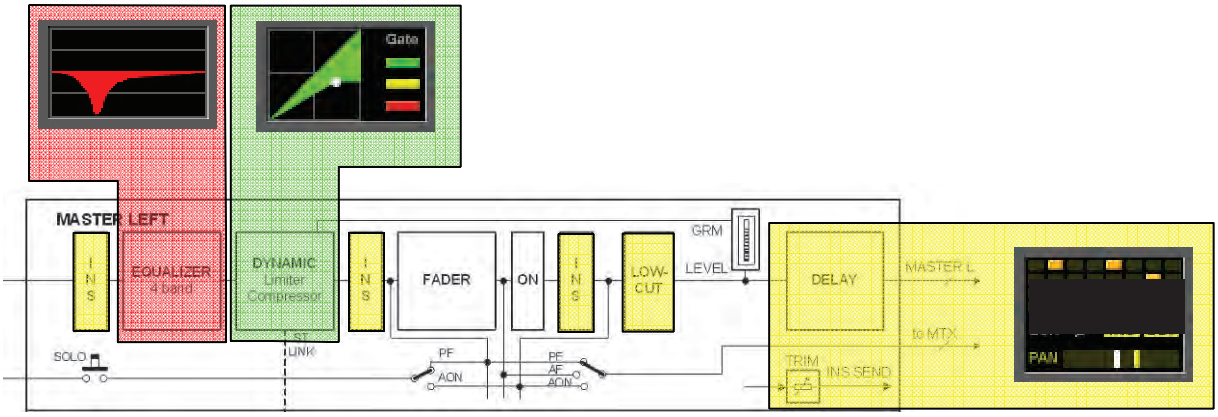


Figure 5-8: The Relationship Between Touch Fields And Master Functions.

EQUALISER

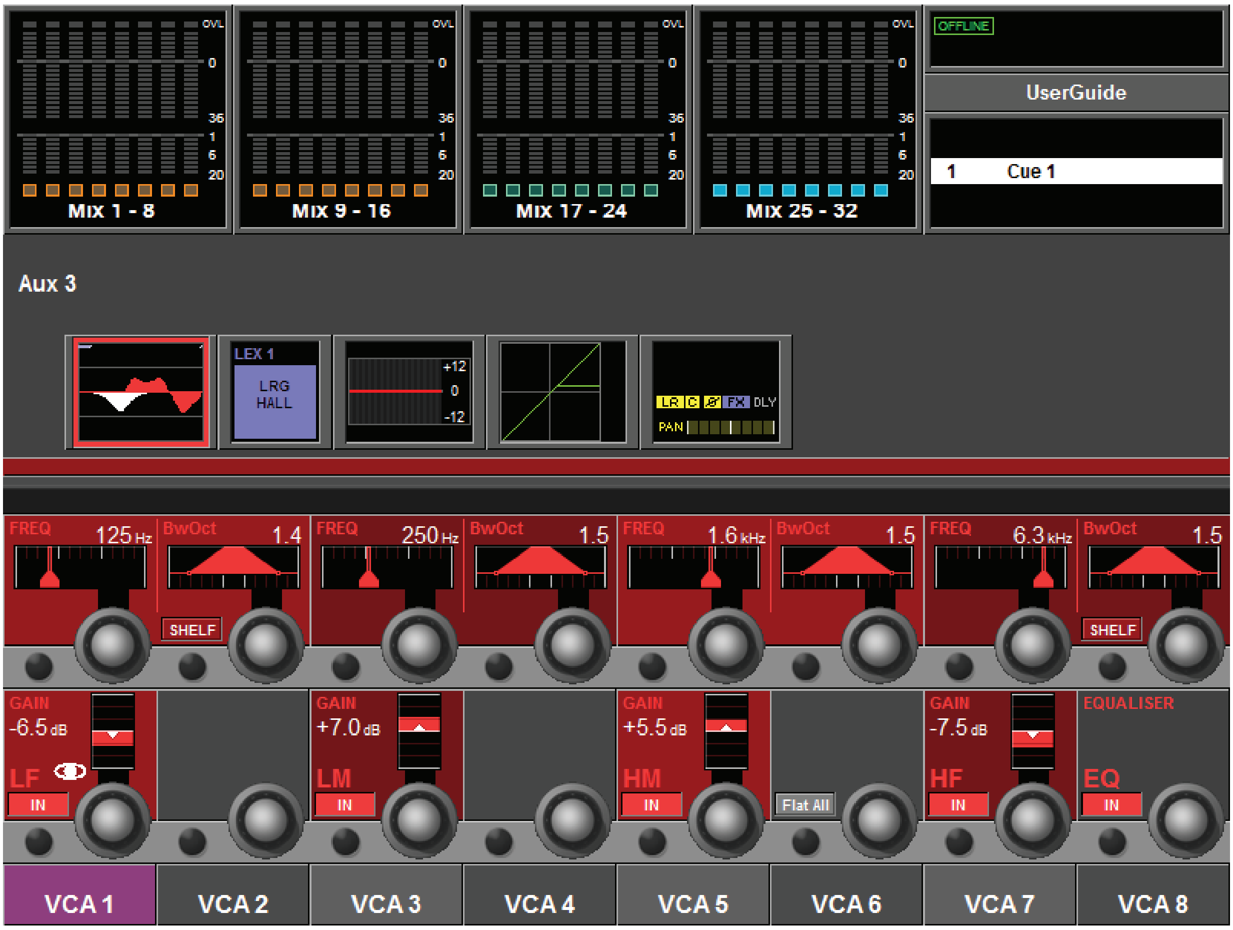


Figure 5-9. Equaliser Controls.

The controls are identical to those of the input channels' EQ.

The VST encoders and keys allow the 4-band parametric EQ to be adjusted and switched in or out of circuit.

DYNAMIC

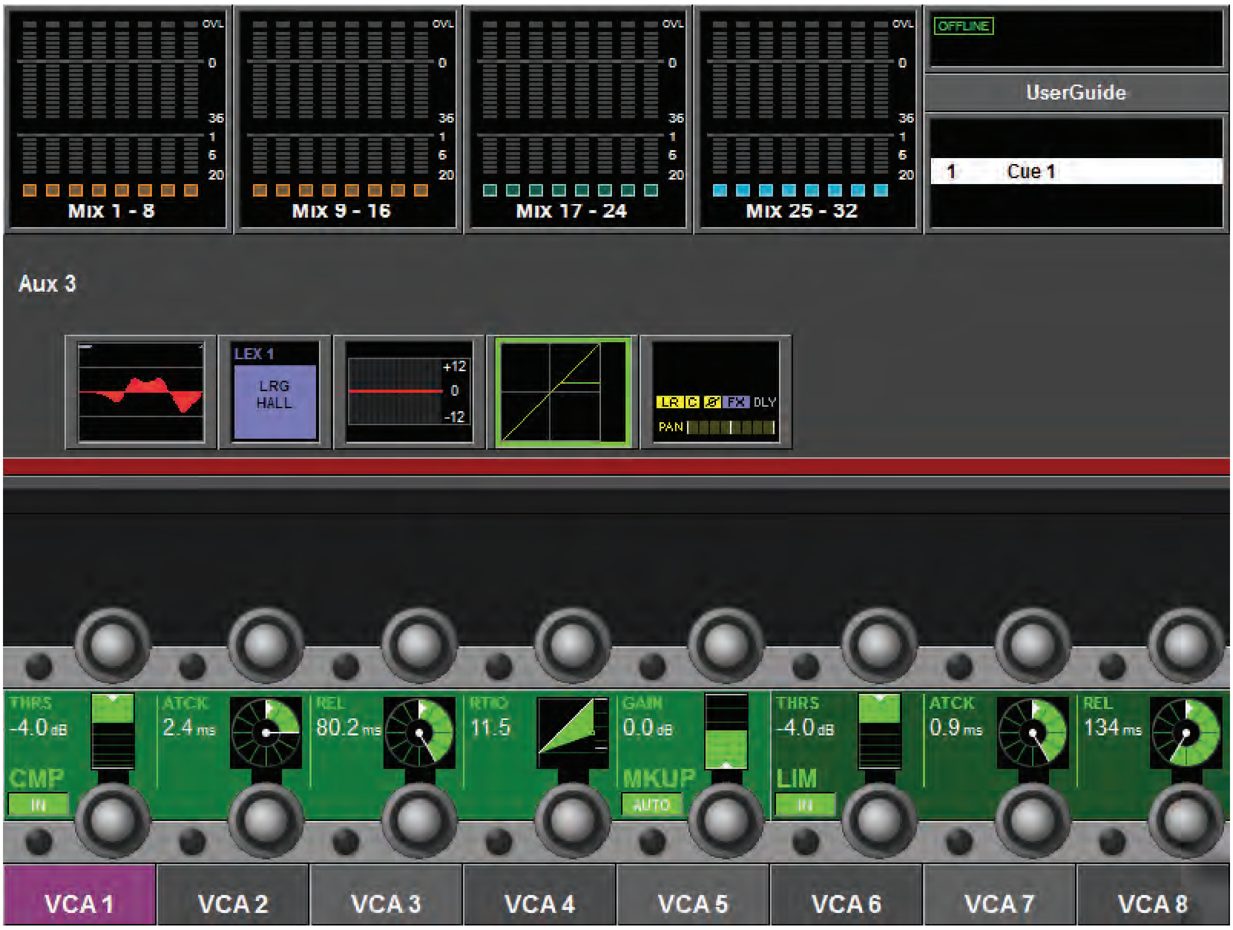


Figure 5-10. Dynamics Controls.

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

PAN

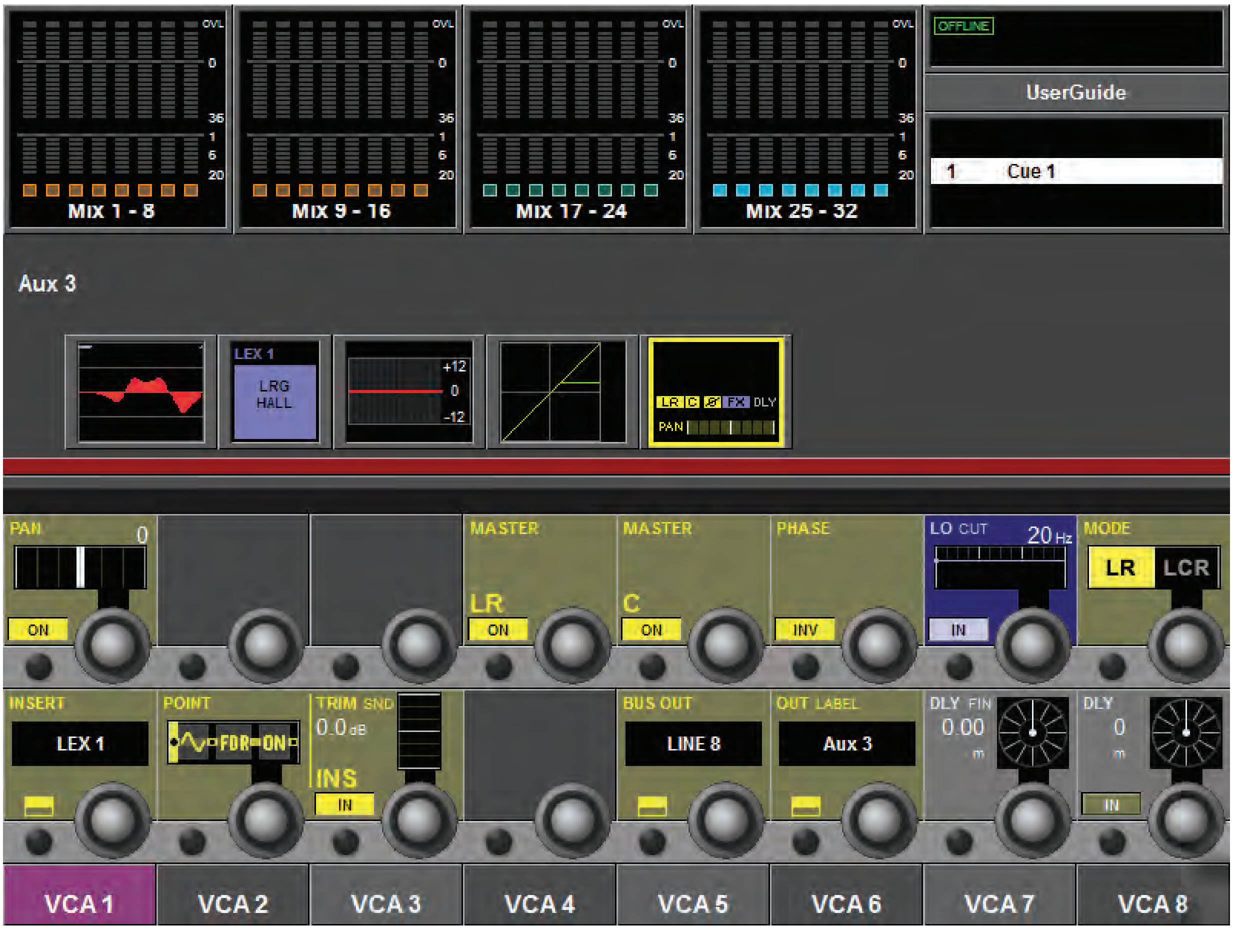


Figure 5-11. Pan.

The controls for the PAN section and Insert section are similar to those of the input channels.

Note that for L, R and C master busses, the PAN, MASTER LR and MASTER C fields are not available.

The extra functions unique to output busses are as follows.

LOW CUT Field

The encoder adjusts the Low cut frequency in the range 20 to 600Hz.
{IN} switches the Low Cut filter in and out.

PHASE Field

{PHASE} inverts the phase at the output.

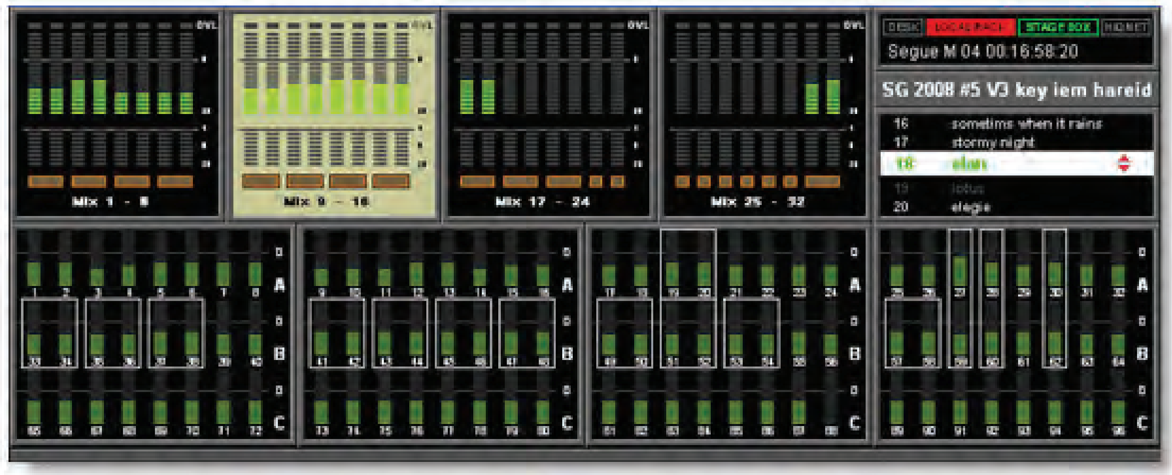
DLY & DLY FIN field

Encoder changes the input delay in milliseconds, metres, or feet & inches. The DLY FIN control allows fine adjustment in 0.02ms/.02 mtrs/0.1” steps.
The DLY control allows coarse control with range of 0.. 100 mS; 0..34mts; 0..112 feet
{IN} enables the delay function.

Output Buss Temporary Activation

Temporary Activation means temporarily assigning a group of Channels or Bus Masters to the far right-hand fader bay of a Vi2, 4, or 6 console.

The main point of this feature is that it gives a faster alternative to using the ALL BUSSES fader page, which switches ALL the Input Faders to Buss master mode



To use this feature, touch the meter overview screen in the block of meters corresponding to the Output busses that you want to access. The relevant section of the ALL BUSSES Fader page will then appear on the far right-hand bay on the surface. The left-hand side of the console will remain in Input control mode.

To switch Temporary Activation off, touch the output meter screen again, or press any Fader Page button to switch to another Fader page.

Mute All Outputs

The Mute All Outputs function allows the whole console to be temporarily muted with one button press. This can be convenient when you need to leave the console unattended, or can be used to prevent unexpected audio output from the console when loading unfamiliar Show files.

To use the Mute All Outputs function, **press and hold** the MUTE ALL O/P button, located adjacent to the main Power button on the console front panel.

The button flashes with red illumination to draw attention to the mute condition.

The press and hold function is intended to prevent accidental operation!

To **unmute** the console, press the MUTE ALL O/P button again briefly (no press and hold necessary). Whilst the console is muted with Mute All Outputs, the input and bus output mutes will show red illumination. The Master LRC mutes will switch off (no illumination) but do not have the red illumination capability. The Monitor Outputs are not affected by Mute All Outputs, this is so that talkback and/or Solo functionality can still be used (eg for Line checking purposes).

The Mute All Outputs button state is not stored in the Show file, which means that the console can be muted and then a different Show loaded without the mute function being cancelled. The console can then be unmuted when it has been established that the audio levels are stable and as expected. (This is not possible by using conventional Mute Groups or VCA Master mutes, unless the Show was previously saved with these muted).

Note that it is possible to use the **Mute Safe** function (accessed from Monitor Setup page, then Setup sub-page) to prevent certain inputs or outputs from being muted when you activate Mute All Outputs (eg to keep a DJ channel running). This function does not exist on the LRC Master outputs however (although you can send the LRC pre-ON to Matrix Outputs to get around this if you need to keep the masters running).

Note: Direct Output from input channels will not be muted by Mute All Outputs, unless they have been set to 'post-ON' in the direct out Point setting on the Input

SIGNAL FLOW

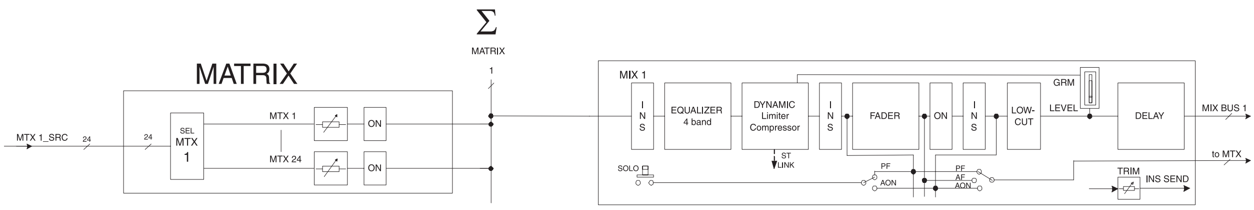


Figure 6-1: Matrix Signal Flow (Vi4 shown, Vi6 has 32 contributions to each matrix, Vi2 has 8).

Only one of up to 16 matrix paths is visible in figure 6-1.

FUNCTION

Instead of a simple output matrix Soundcraft Vi Series™ has a built-in freely-configurable matrix that can have up to 16 Outputs (mono).

Each matrix output is a mix of up to 8 (Vi2), 24 (Vi4) or 32 (Vi6) configurable sources and contains full processing including Equaliser, compressor/limiter and Delay.

Sources to the Matrix can be signals from busses, channel direct outputs or inputs.

Hint: The sources for each matrix out can also be individually patched, the matrix can be utilized as 16 different mixers, each with up to 8 (Vi2), 24 (Vi4) or 32 (Vi6) inputs, with output processing. This could be used, for example, to send the same basic aux mix to several musicians, with each also getting a blend of their own input source mixed in.

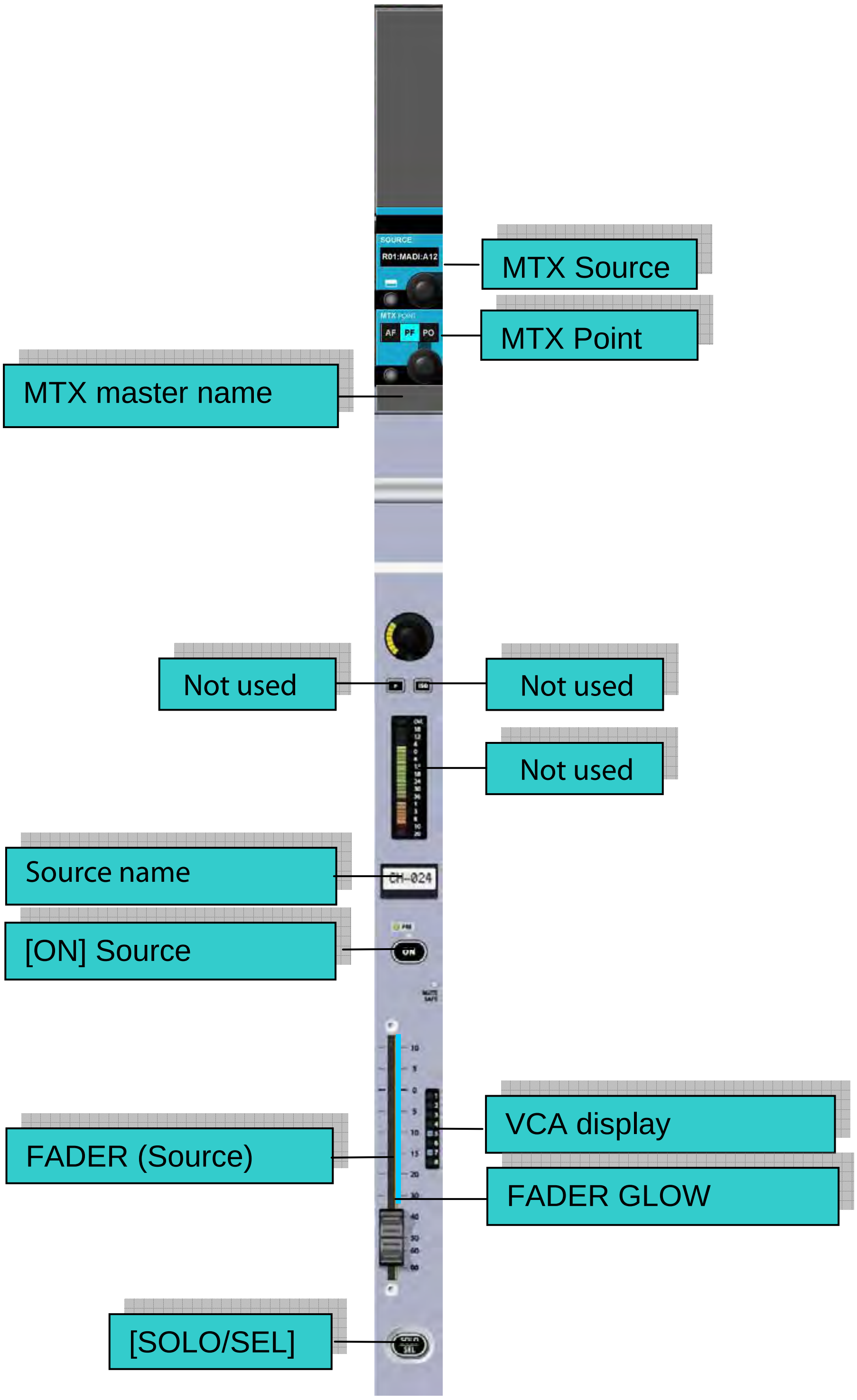


Figure 6-2: Matrix Strip.

Adjusting A Matrix Send Level

Ensure that the [ALL BUSSES] key is **NOT** selected.
Press the [SOLO/SEL] on a matrix master. All the input bays will change to the matrix contributions view and the Faderglow™ illuminates with the matrix colour. The input strip [ON] keys act as ON for the matrix send signals. The fader adjusts the contribution level of the desired source (1..32).

MATRIX CONFIGURATION

This screen opens on all 4 input bays after pressing SOLO/SEL on a matrix master.

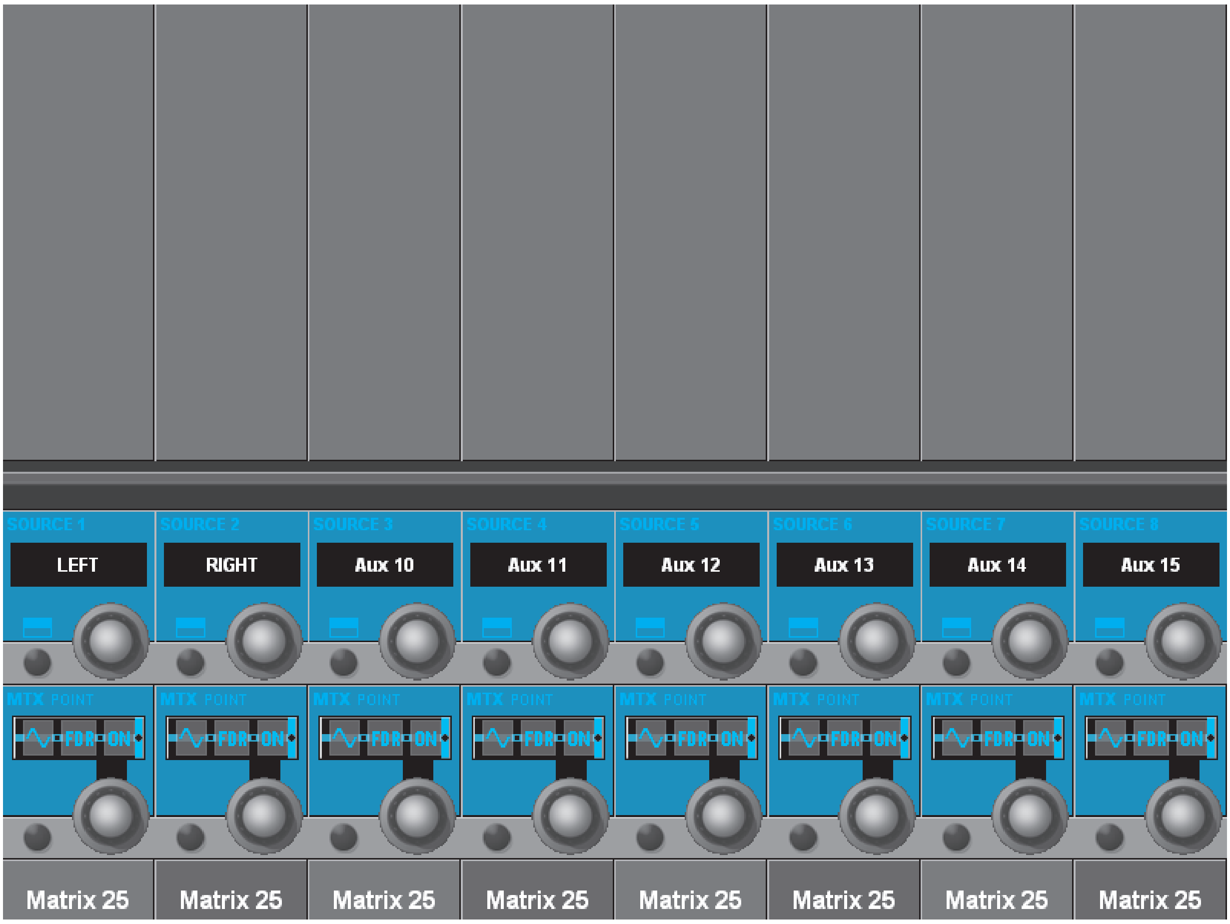
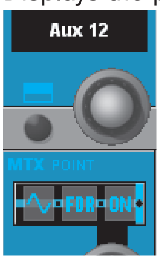
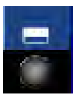


Figure 6-3:Matrix Configuration Page

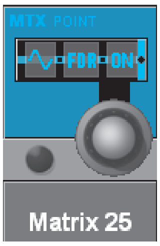
SOURCE field

Displays the patched source.



The  key opens the Matrix patch configuration page - see chapter 11.

MTX Point Field



The encoder changes the point from where the matrix source signal is taken.
Which points are available depends on the Source selected.

MTX Source	Input	Pre-Filter	Pre-Processing	Pre-Fader	Post-Fader	Post ON
Input	*					
Direct OUT		*	*	*	*	
Master BUS				*	*	*

GENERAL INTRODUCTION

Each input channel strip contains three encoders (see Figure 7-1A): encoder 1 & encoder 2 are located in the VST fields in the lower screen area, 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.

The master section has 16 VST encoders and 4 panel-mounted encoders with LED rings (see Figure 7-1B): the TB/OSC Level Control encoder, and the Solo Blend, Solo Trim & Phones Volume encoders. These last four are dedicated to their respective functions.

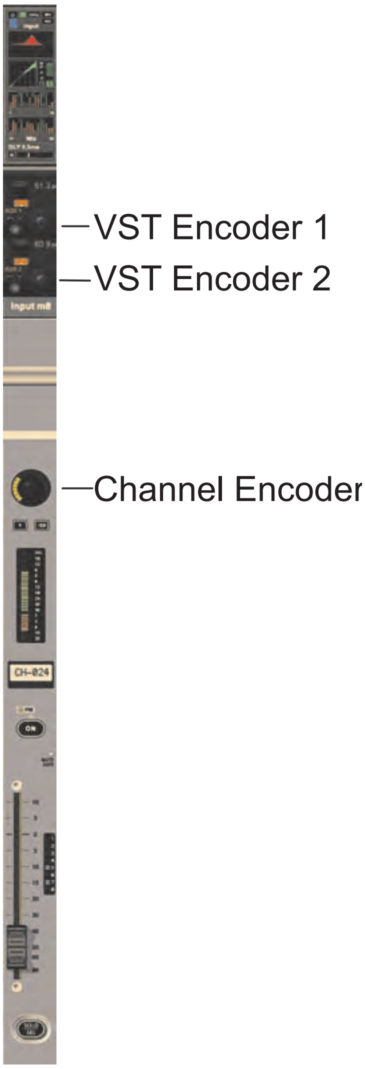


Figure 7-1A: Input Channel Strip Encoders.

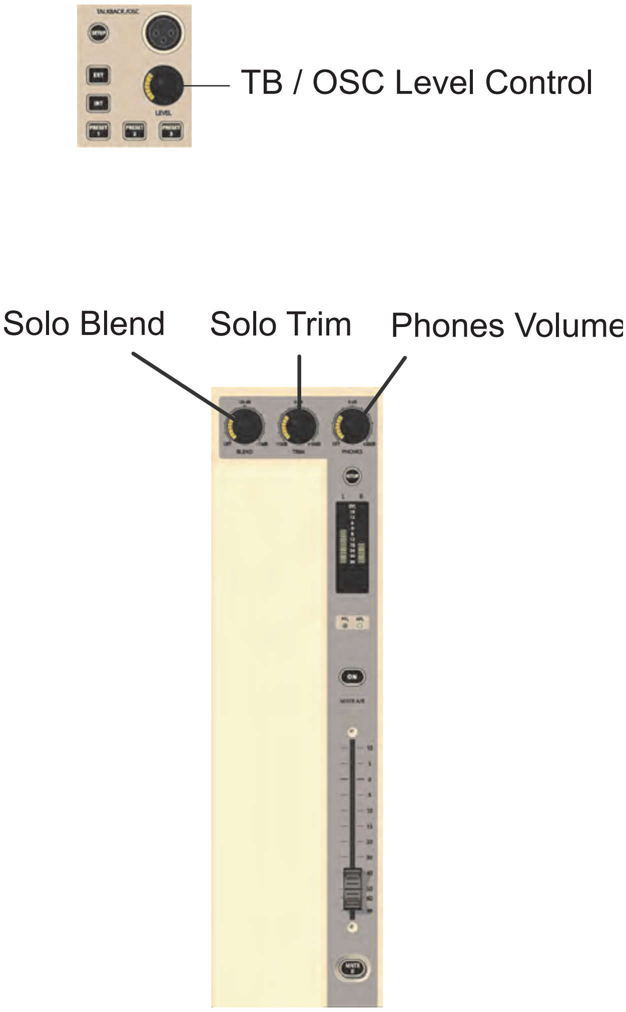


Figure 7-1B: Master Section Panel Encoders.

CHANNEL ENCODERS

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] and [PAN] keys on the Encoder Mode panel (see Figure 7- 2).

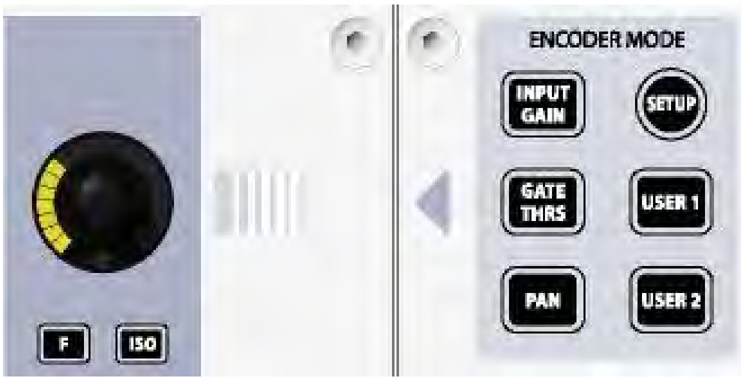


Figure 7-2: Encoder Mode Panel.

[USER1] and [USER2] are currently used to set the channel encoders to control the Low-cut Filters and the High-cut Filters respectively.

[SETUP] is reserved for future use.

HINT: If [ALL BUSSES] is active or a MATRIX output is soloed, the channel encoders are disabled and have no function (the previous function is remembered however).

HINT: If Stereo Aux sends are assigned to the channel faders (using FLW Output Solo), the channel encoders control the PAN for their channel’s contribution to the soloed Stereo Aux bus.

CHANNEL VST ENCODER 1& 2

Figure 7-3: Default VST Encoders 1 & 2 For Input Bay.

VST Encoder Priority

If a higher priority assignment action occurs, VST encoder functions change immediately to this mapping. The priority order of the possible assignment actions is:

Priority Function

highest	CHANNEL EXPANDED FUNCTION (e.g.: EQ, Dynamics, etc)
	FOLLOW OUTPUT SOLO
	FAST ASSIGN
lowest	USER or DEFAULT

HINT: If any touch field is activated, the 16 VST encoders are assigned with expanded channel function parameters (see Chapter 4 for full details).

By default, the VST encoders 1 & 2 are assigned as AUX 1 and AUX 2 send level controls for their input channels.

Figure 7-4: Vistonics Mode Panel

Changing Encoder Function

The function assigned to the VST encoders can be changed by the user via the Vistonics Mode panel (see Figure 7-4).

The two [FAST ASSN] keys provide 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 (there will be no influence on audio, the Output solo is not activated). The relevant row of VST encoders will now be assigned to that Output master, and the [FAST ASSN] key in question will illuminate.

Hint: Only Output Masters which are configured as Auxes use the encoders. Group Masters do, however, use the VST button next to the encoder.
HINT: Fast assign mapping is removed by pressing/releasing [FAST ASSN]. Fast assign always works GLOBALLY for all input strips

The [SETUP] key is reserved for future use.

If [USER] is pressed the two VST encoder rows will be assigned as AUX 3 and AUX 4 send level controls for their input channels. The [USER] key is illuminated when active.

The two [FLW] keys activate the FOLLOW SOLO function for their respective encoder rows. This means that pressing an Output Solo/sel will automatically assign the soloed Output to this row, overriding the default or the [USER] layer. Note that only one [FLW] can be active at a time.

The [PAN] key, which is illuminated when it is active, only has an effect on Aux Masters which have been configured as stereo pairs. If such a pair is assigned to a VST encoder row, and if the [PAN] key is active, the encoders will control the pan between the pair rather than the contribution level.

HINT: 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.

The [PRE/POST] key isn't used for any encoder functions, but for the sake of completeness its function is described here. The [PRE/POST] key allows the user to configure Aux sends from channels, when they are assigned to the two VST encoder rows, as pre or post-fader.

MASTER BAY PANEL ENCODERS

The master bay has four panel-mounted encoders with LED rings (see Figure 7-1B): the TB/OSC Level Control encoder, and the Solo Blend, Solo Trim & Phones Volume encoders. These are all dedicated to their respective functions. A description of their functions is given in chapter 9 of this manual.

MASTER BAY VST ENCODERS

Figure 7-5: Default Master VST Encoder Assignment.

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 (see Figure 7-6). 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 paticular touch-area is touched, the VST encoders are assigned to appropriate expanded functions. These functions are all described in detail in chapter 5 of this manual.

HINT: [METER LOCK] must not be enabled, otherwise the Processing Area will not be displayed. If multiple Output Solos are activated, the parameters for the LAST soloed Master are displayed.

Figure 7-6: Master VST Mode and Switch Function Panel.

Master Vistonics Switch Function Panel

Although the VST switches which are located next to the encoders are not the subject of this chapter, for the sake of completeness a note on assigning their function is included here.

Using the three buttons on the Switch Function panel, the VST switches can be assigned to [TB ASSN], [ON/OFF] or [SOLO/SEL]. [SOLO/SEL] is the default. The assigned function is the same for both Page A and Page B.

These VST switch functions are, of course, over-ridden when the encoders are assigned to expanded functions.

GENERAL

Layers, or Fader Pages, allow the user to access different views of the DSP channel structure of the mixer, on the control surface. There are fixed, and assignable layers.

INPUTS

The Control Surface for the Soundcraft Vi Series™ contains three (Vi4) or four (Vi6, Vi2) fully equipped Input Bays that allow direct access and state information overview for 32 (24 on Vi4, 8 on Vi2) Input Channels.

Therefore two layers are required to allow operation for all 64 inputs, or 3 layers for 96 input channels.

Changing Layer is done by simply pressing the desired Input Fader Page key [A] or [B] or [C].

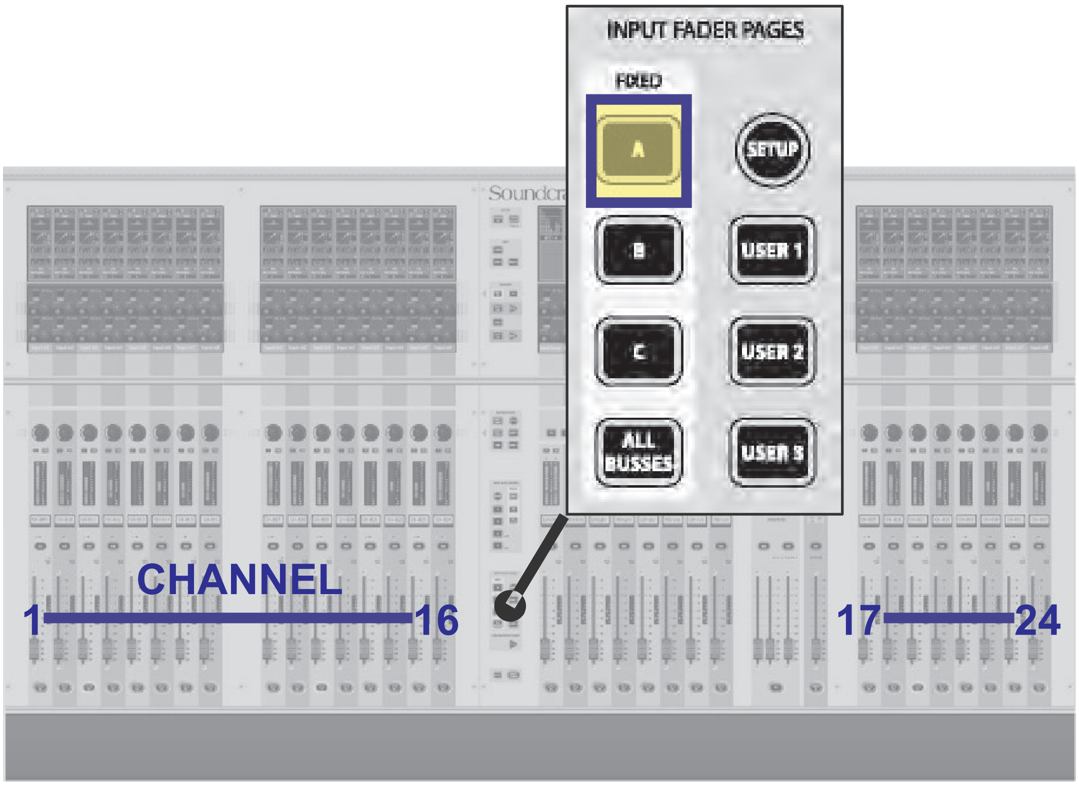


Figure 8-1: Desk In Layer A operation.

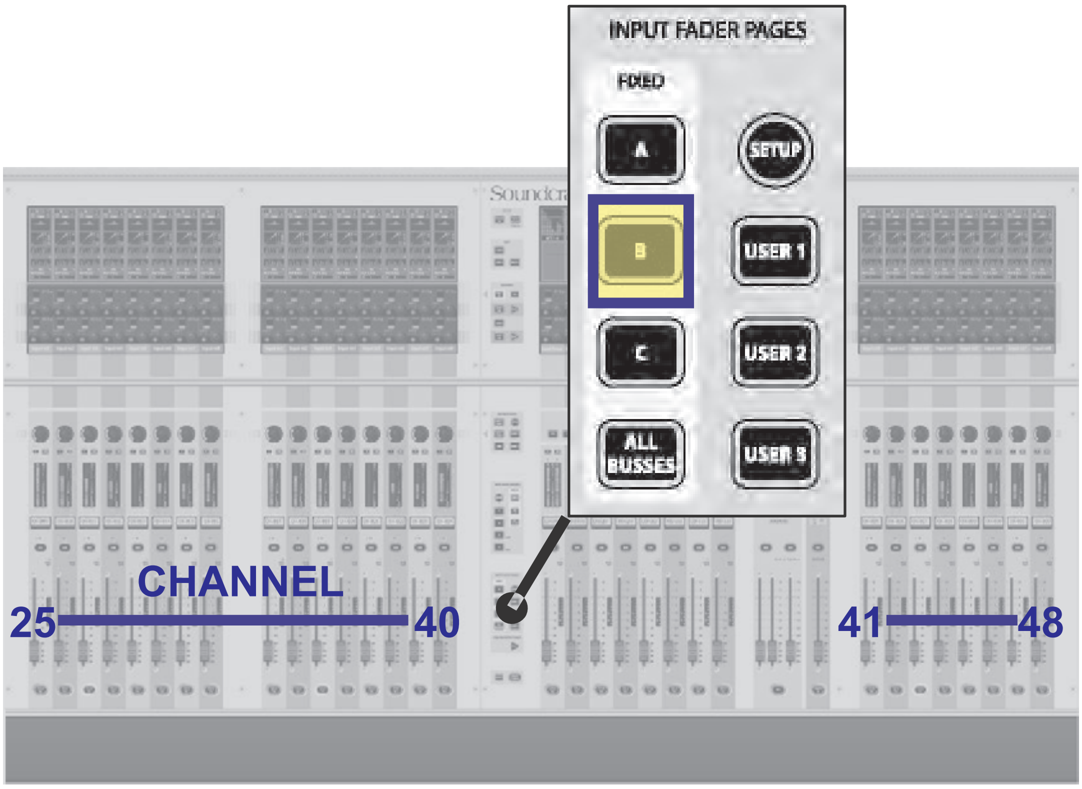


Figure 8-2: Desk In Layer B Operation.

Temporary Layer

Touching an input meter field on the Master Vistonics ™ Screen will temporarily display the 8 channels that are in this group on input bay 4 of the surface.

The input fader page selector will blink to indicate that the bay 4 is temporarily remapped. Leaving the mode can be done by pressing the input fader page selector or by touching the meter panel field again. A highlighted box appears around the 8 selected meters when they have been touched.

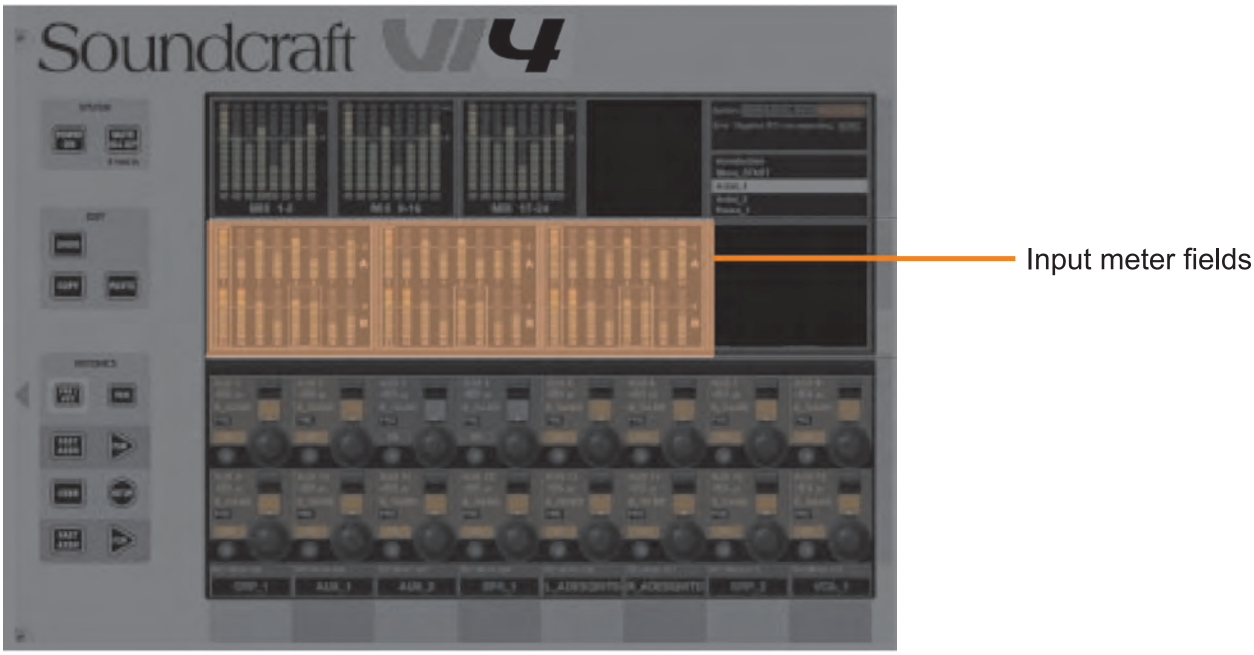


Figure 8-3: Input Meter Fields.

OUTPUTS

Bus Masters can be accessed with the master fader strips in the VST Master Area or with the ALL Bus layer on the Input Bays.

Master Fader section

There are 6 layers for the master bay: VCA and A to E. They map the following master faders to the master bay as follows: VCA maps VCA 1-8, A maps busses 1-8, B maps busses 9-16, C maps busses 17-24, D maps busses 25-32, E maps VCA 9-16. In the factory-default Front-Of-House Show, busses 1-16 are set as Aux, busses 17-24 are Groups, and busses 25-32 are set to Matrix.

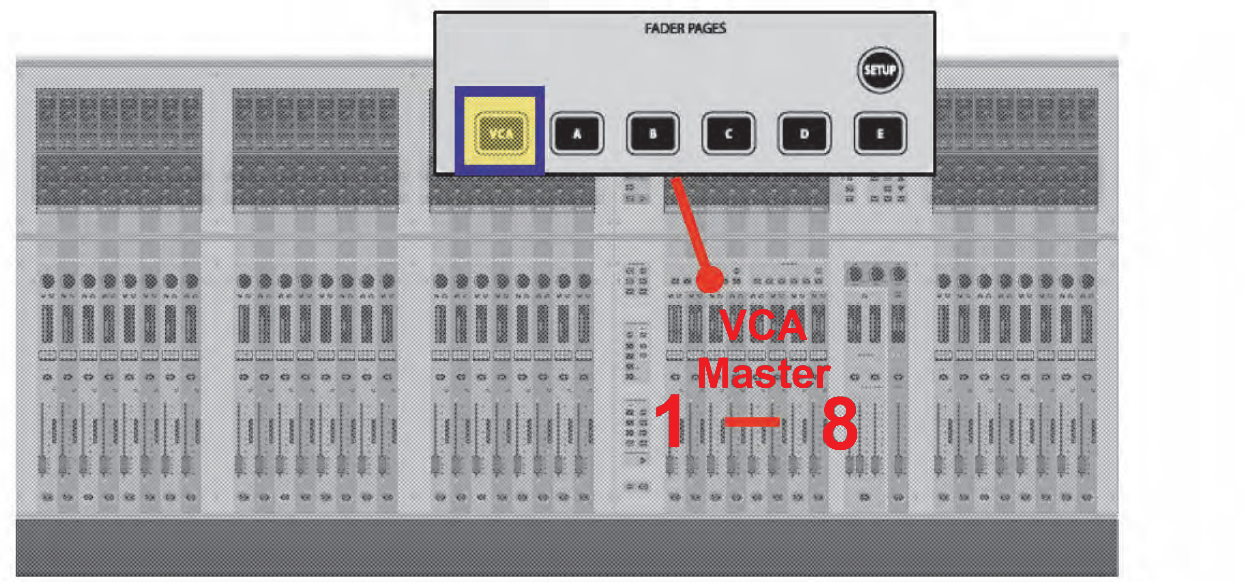


Figure 8-4: Master Bay In VCA Operation.

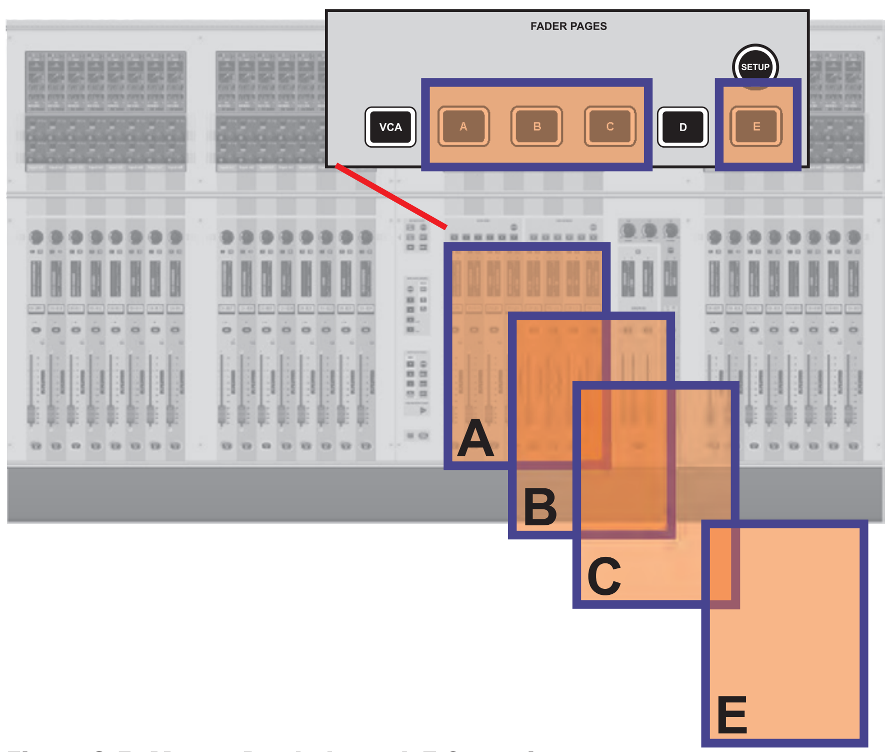


Figure 8-5: Master Bay In Layer A-E Operation.

ALL Busses

If [ALL BUSSES] is selected in the Input Fader Pages, all 32 Busses will be assigned to the four Input Fader Bays. This Layer allows a quick way to compare the outputs or quickly change the processing of the Busses.

HINT: Configuration of the BUSSES, e.g. Format (mono/stereo) and Type (Aux, Grp, Matrix) is also handled in this Layer.

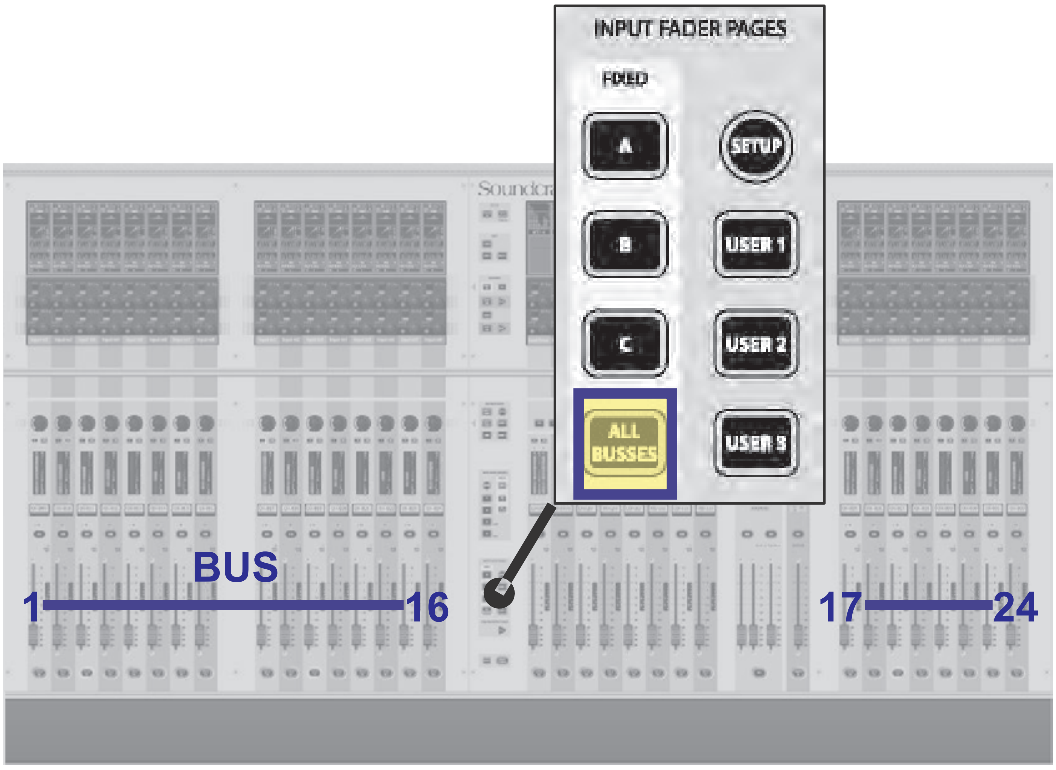


Figure 8-6: ALL BUSSES Operation.

VST Master Area

The VST Master Area allows all 32 Bus Masters to be accessed even if the 8 output faders are assigned to control VCA Masters.
There are two layers of encoders: Page A displays busses 1-8 and busses 9-16, Page B displays busses 17-24 and busses 25-32. In the factory-default Front-Of-House Show, busses 1-16 are set as Aux, busses 17-24 are Groups, and busses 25-32 are set to Matrix.

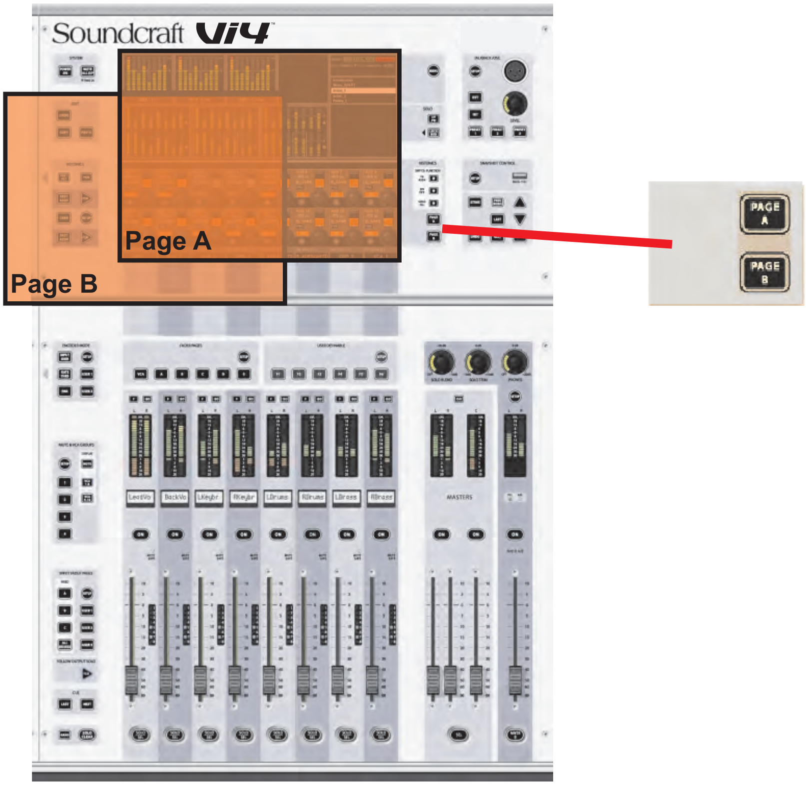


Figure 8-7: VST Master Area.

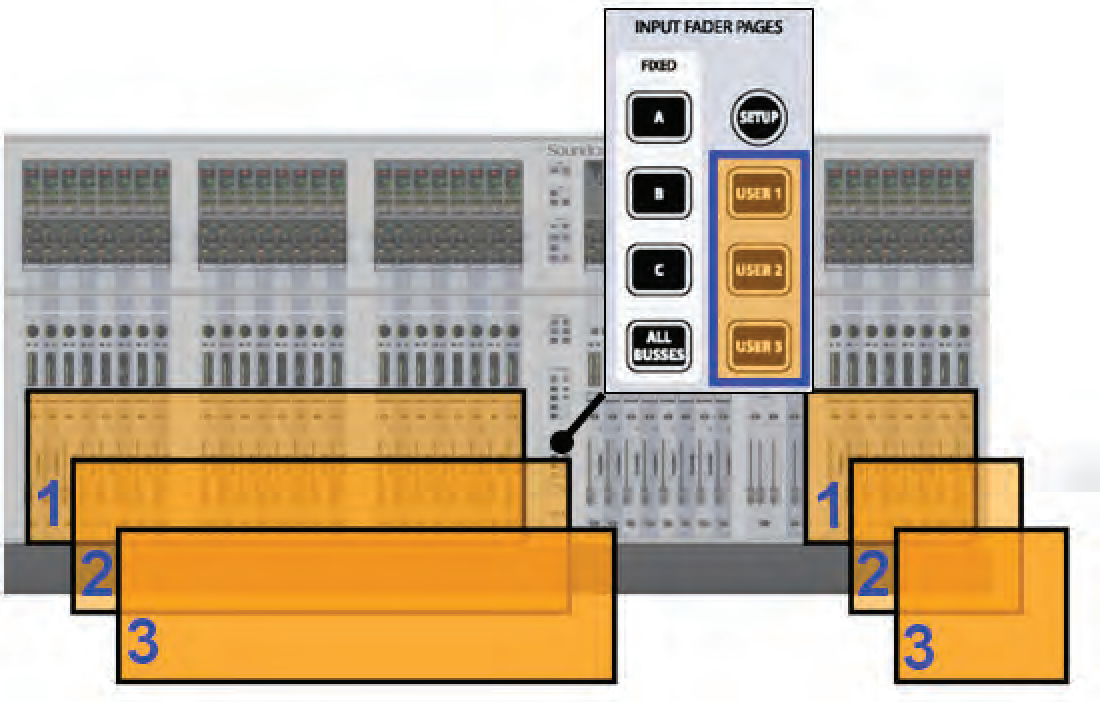
User-defined Fader Pages

What is the difference between 'Fixed' and 'User-defined' Input Fader Pages?

'Fader Pages' are defined as arrangements of channel strips on your console. Up until Version 4.0 software, only 'Fixed' Fader Pages (A,B & C) were available on Vi consoles. These fixed pages are simply the input channels 1-32 (for example on Page A of a Vi6) arranged in sequential order in the same way as channels would be on an analogue desk.

With User Defined Fader Pages, nothing changes as far as the Fixed Fader Pages go, but what we add is the ability to take the channels that exist on those fixed pages, and have them available again on three special pages but in an order that can be chosen by the user. Changing the order here does not affect the signals patched into them or any of the processing or labels that might already have been applied – everything moves with the channels. Re-arranging the channels on User-defined pages also does not affect the positions of those channels in the Fixed Fader Pages – you can always go back to the Fixed Pages at any time and find the channels in their original sequential positions.

There are three User Defined pages User 1, 2 & 3 that can be created, and each of these can contain any combination of the channels that appear on the three Fixed layers. It is also possible to arrange VCA Master faders within the User-defined Fader pages, along side input channels. There are no restrictions on how many times you can use a particular channel, so for example it is possible to assign a vocal channel to the same fader in all three User Pages, meaning that it will appear to remain in the same place on the surface regardless of which User Page is selected.

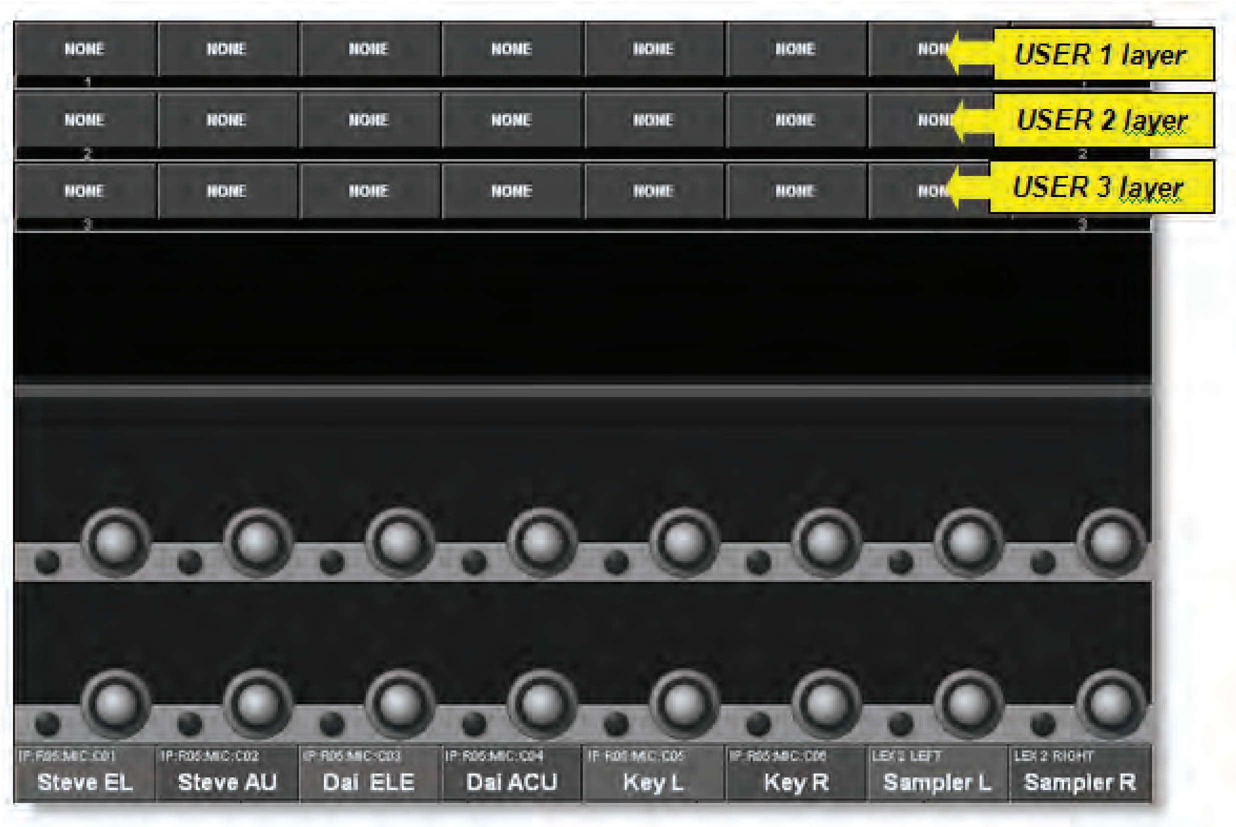


Creating User Defined Input Fader Pages

Step 1 – Open the Setup screens

Press the Setup button in the Input Fader Page control section.
This opens up identical Setup screens across all of the console's input Vistonics screens – see the picture below.

Each Setup screen contains three rows of buttons, corresponding to the User Pages 1, 2 and 3 for that bank of 8 fader strips. If you load a Default Show, all of the faders in all of the layers will have a default setting of NONE, which means that no channels are yet assigned to any of the faders in the three User Pages *(the exception to this is the Vi2 – this has channels 25-32, 33-40 and 40-48 pre-assigned to the User Pages 1-3).*



Hint: A User Layer which has all the fader strips set to 'NONE' will result in black screens across the whole console if you select this User page by pressing its Fader Page button after switching off the Setup mode. This is the normal state of the User 1-3 pages in the console's Default shows, the idea being to provide a 'blank canvas' on which can be assigned your required channel layout.
If you load a show that has had User Layers already programmed, then you will see these User Layers when you press their Fader Page buttons, because the User layer setup is stored with each Show file.

Step 2 – Open the Channel Select screen

To start assigning channels to the fader Strips, touch any of the buttons labeled ‘NONE’ in the previous screenshot, corresponding to the position and layer of the fader you want to assign something to. Normally you would start at the top left and work across each layer, assigning the faders in order.

Touching any of the ‘NONE’ buttons opens up the ‘Channel Select’ screen that then allows you to choose any of the input channels on the desk to be assigned to your chosen fader:

The tabs on the right side of the screen allow all available input channels to be accessed (the last tab 65-96 will only be shown if you have a Vi4 or 6 that has the 72ch/96ch DSP upgrade fitted).

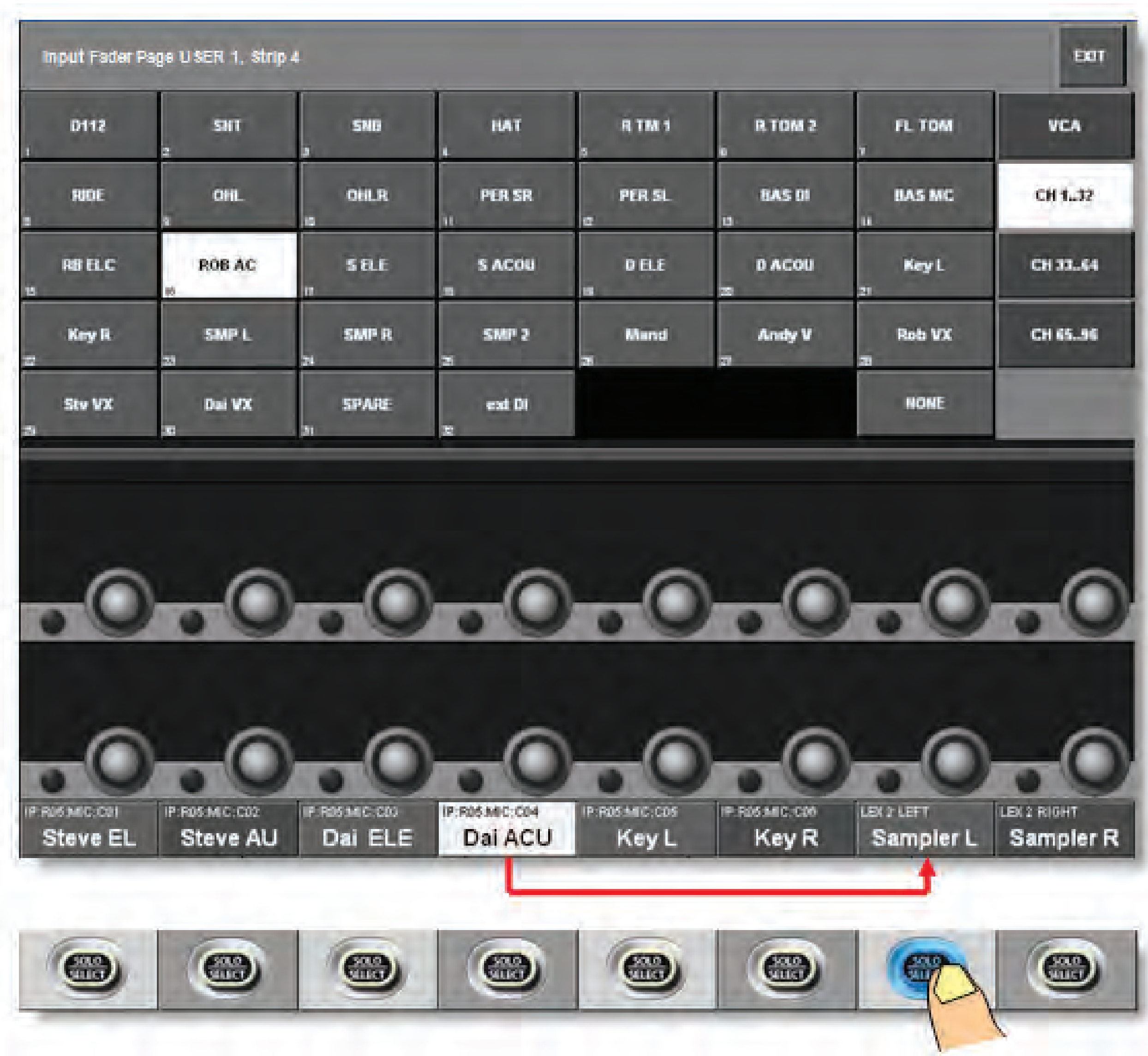
Each channel select button shows the channel’s ‘short label’ name in the centre of the button, and the channel’s number, corresponding to its position on the fixed layers A/B/C, in the bottom left corner.

In the case shown above, the channel ‘ROB AC’ is being chosen to be the assignment for Strip 1 on User Layer 1. As soon as you select your channel, the select screen will automatically close and return to the Setup page, and you will be able to see your assigned channel on the first strip of User layer 1, like this:

Assigning VCA Master Faders to Input strips

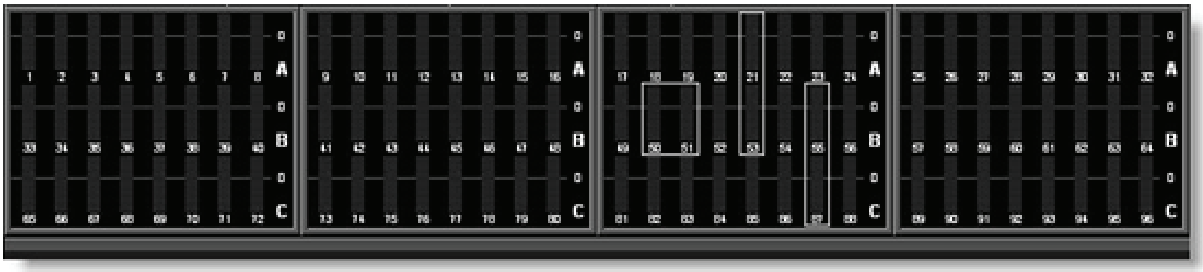
As well as choosing input channels to assign to fader strips on the User Layers, it's also possible to assign VCA masters alongside the inputs. The 16 VCA Master faders are displayed in the channel select page by touching the VCA tab on the top right of the screen:

Hint: It is also possible to leave the channel select screen open, without selecting an channel or VCA, and use the Solo/Sel buttons on the fader panels below the desk to scroll to a different fader strip to the one you started with, before selecting the required channel:



Paired Input channels within User-defined Fader Pages

The pairing system on the console allows channels to be paired either horizontally within any of the fixed Fader Pages, or vertically between any two adjacent fixed Fader Pages. The pairing indication is summarised using white rectangles on the console's input meter overview screen, the layout of which corresponds to the fixed Fader Pages A, B and C.



It is **not** possible to create pairings between channels that are adjacent either horizontally or vertically, on the **User-defined** fader pages. This is because channels can be placed in any order within the User-defined fader pages, and if pairing was allowed, it would be possible to create links between channels which might not be adjacent on the Fixed fader page (eg ch's 1 and 64), and therefore could not be shown on the meter screen –and could not be handled by the console's DSP structure.

You can however still **view** or **edit** the pairing of channels from within a User-defined Fader page – the standard pairing screen is still available within the Input function block of an Input channel strip.

What you will see when you open the pairing page from an Input channel is the same screen as you would see if you opened the pairing page of that channel from within a fixed Fader page – ie the pairing choices you will be offered by the page will be the channels that are adjacent on the fixed fader pages to the one you have selected.

So let's say you have Fader Page USER 1 active, and on this User page you have channel 37, with channel 1 on its left and channel 64 on its right.

If you open the pairing page for Ch37, you will see the screen below, which is the same screen you would see if you opened the pairing page for Ch37 from a fixed fader page – ie: it has Ch36 on the left and Ch38 on the right.

Be aware therefore that you will see channels laid out as they are in the fixed fader pages, if you open the pairing page from a User-defined fader page.

Stereo Input Channels in User-defined Input Fader Pages

Stereo paired channels are treated as if the two halves were separate mono channels, as far as assigning the channels to User-defined Fader pages goes.

In other words it is not necessary to assign both sides of a stereo pair to a User page, if you want to maximize the use of faders.

On Fixed Fader pages, two types of indicators are used at the bottom of the touch screen to show that channels are paired, and how they are paired:

Vertical Horizontal

When you assign only one side of a Stereo pair of channels to a fader in a User-defined Fader page, you will see a slightly different indication, as seen in the channel strip on the right.

The two ‘gear wheels’ without any white line in the blue strip below the Pan display indicates that this is one half of a paired channel (note the difference with the vertical and horizontal indicators shown above). The other half of the pair may be adjacent to this channel on the User page, or it may not be assigned at all on this User page.

It is important to remember however that not all parameters are linked when you create paired channels – most of the input stage such as gain, trim, phase etc, plus the pan controls are not linked, and so it will only be possible to adjust one side of these from the User Fader page, if only one fader from the pair is assigned.

However, remember that all channels including any ‘dropped’ halves of stereo pairs are always available in their original locations via the Fixed pages A/B/C or via the temporary right hand bay activated from the Input Meter screen.

User Defined Output Fader Pages

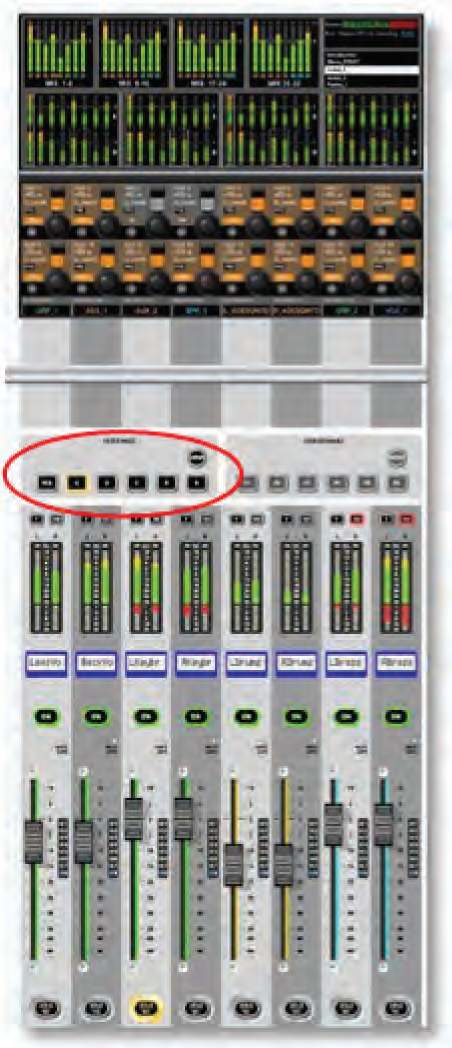
With previous software versions on the Vi consoles, the Output Faders in the centre section of the console have only been accessible via 4 ‘fixed’ fader pages A,B,C,&D, with two fixed VCA fader pages under ‘VCA’ and ‘E’. Version 4.0 software unlocks the page buttons A – E and allows flexible configurations of Output busses and VCA Masters on any of the pages. (The ‘VCA’ page remains fixed and is dedicated to VCAs 1-8).

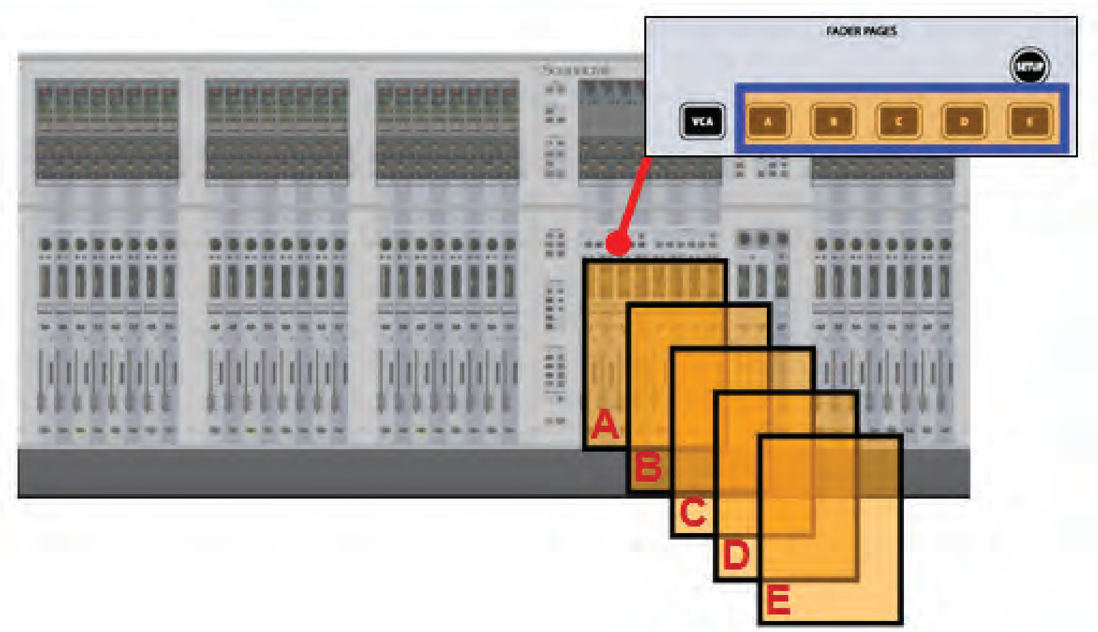
Although most users are quite happy with the fixed fader pages on outputs, because it’s generally easy to find things when they run in banks of 8, there are two primary reasons why this new functionality is useful:

* Being able to combine VCA masters alongside audio bus masters in the same fader page. This would be useful if you had a VCA that was controlling several outputs for example.

* Being able to ‘drop’ one side of Stereo bus masters from a Fader Page in order to allow more Stereo outputs to be controlled at the same time.

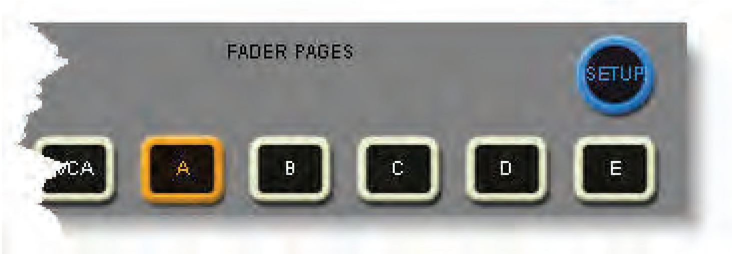
The second of these benefits will be particularly useful for monitor engineers, who will now be able to control up to 8 stereo mixes at the same time without having to change fader page.





Creating User Defined Output Fader Pages

Step 1 – Open the Setup screens

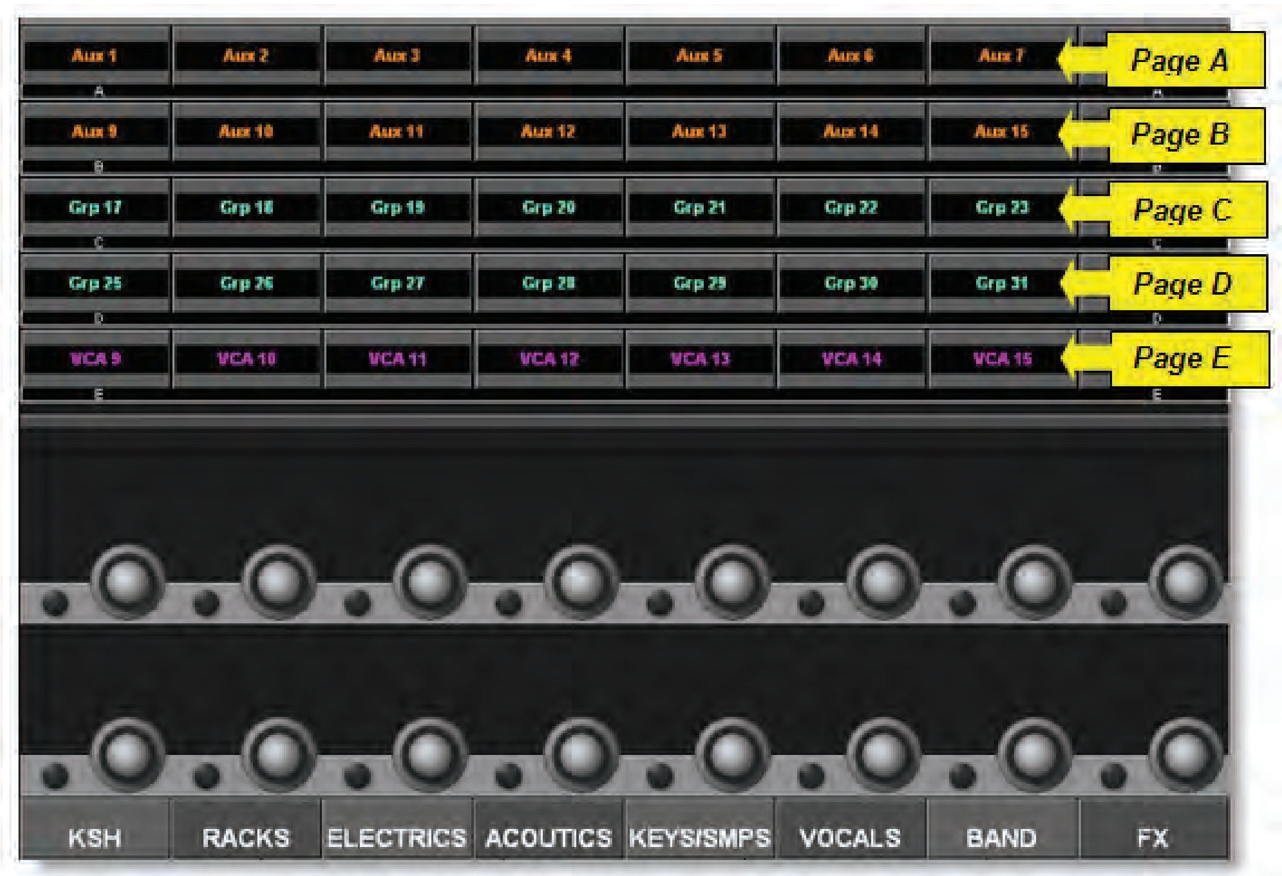


Press the Setup button above the Output Fader page switches in the centre-section of the console.

This will open a Setup screen on the master Vistonics screen – see picture below.

The Setup screen contains five rows of buttons, corresponding to the five Output Fader Pages A – E, for the 8 Output fader strips.

The default setup for these Fader pages will be already familiar – it is the 32 busses of the console arranged in banks of 8 in a sequential order, and with VCA 9-16 on the last page.



You can now change the layout of any of these five pages, if necessary, and store that setup with your show file.

The method is similar to assigning Input Fader Pages, and is described on the following page.

Step 2 – Open the Output Select screen

To change the default page assignment, touch any of the bus names on the Setup page to open the Output select screen.

This screen shows a group of all the available busses and VCA Masters on the console, with the currently assigned one highlighted in white. The Output fader strip that you are changing is indicated by the white highlight on the long name display at the bottom edge of the Vistonics screen. In this example we are changing the assignment of Strip 4 ('ACOUSTICS'), on Output Fader Page C:

The tabs on the top right of the screen allow you to choose either the 32 Output bus masters or the 16 VCA Masters for your selection. You can also choose to have no assignment on a particular fader strip, by touching the 'NONE' button.

Combining Stereo Bus Masters onto a single Master Fader

Prior to V4.0 software, the eight centre-section bus master faders could only control a maximum of 4 stereo busses on one fader page, because the stereo bus masters always used two faders each. The Vi surface however was designed to allow stereo busses to be controlled by single faders, which is why there is a stereo meter above each bus master fader, and Version 4.0 software now allows this mode of operation.

In order to assign a stereo bus master to one fader, it is simply necessary to remove the other fader in the pair from the output fader page, and assign something else to the adjacent fader (eg one fader from another stereo bus).

To do this, first set up the bus structure that you require – eg busses 1-16 set up as 8 stereo Aux busses – then use the Output Fader Page Setup as described in the previous pages, to assign only the odd numbered bus masters to the fader page you require.

The screenshot below shows how the assignment would look for Output Fader Page A:

Setting up Output Fader Page A like this will give eight stereo Aux masters controlled by single faders. Fader Page B has been assigned as 'empty' in this example, but you could assign other things here.

Since all of the audio parameters of stereo bus masters are stereo linked, it is not a problem to 'drop' the even numbered faders like this.

Once you have programmed User-defined Output fader pages in this way, you can still if necessary view the busses in the way they were originally displayed – in sequential order – by one of the following methods:

* Switch to the ALL BUSSES page. The ALL BUSSES page always shows all 32 output busses, regardless of any User-defined setups you have made on the centre-section.

* Use the new Temporary Output Meter Activation feature to assign sections of the ALL BUSSES page to the right-hand input bay, whilst still keeping input faders on the left of the centre-section.

Soundcraft Vi Series™ supports up to 8 MUTE Groups (MG) and 16 VCA masters.

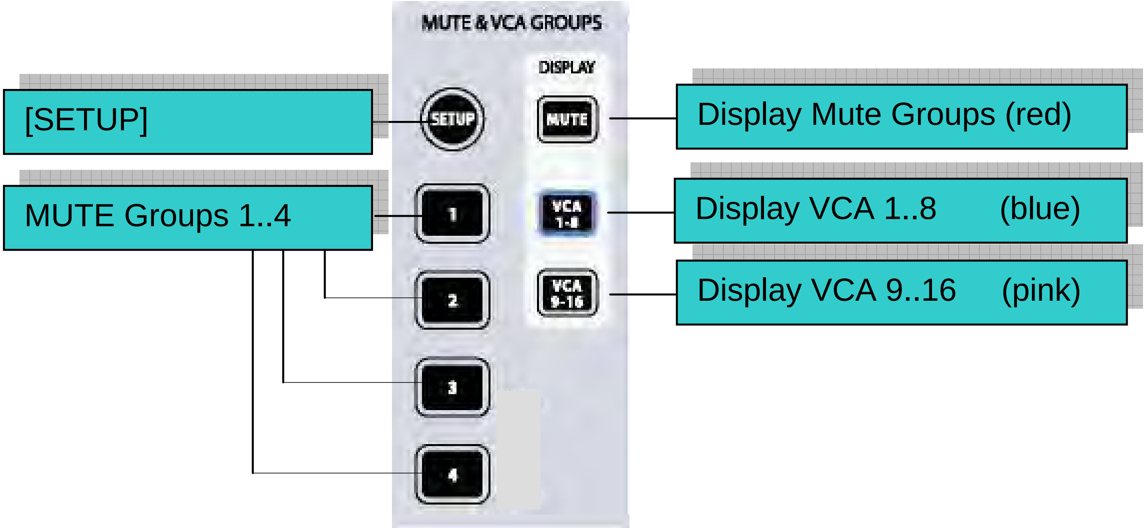


Figure 9-1: MUTE & VCA Front Panel Keys

VCA/MUTE GROUP INDICATION

Each input and output strip contains a VCA/mute group display. [MUTE], [VCA 1-8] and [VCA 9-16] set the display mode for all strips.

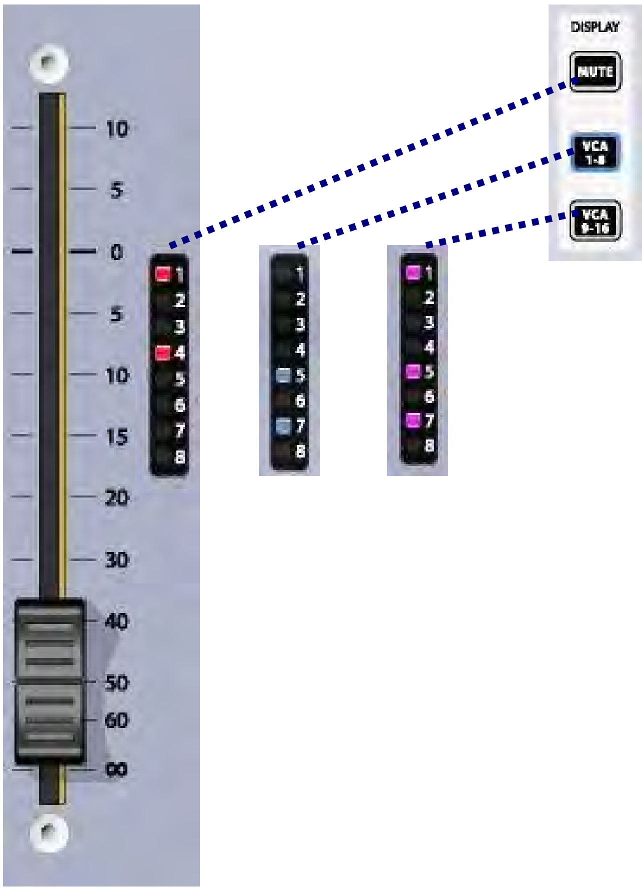


Figure 9-2. VCA/MUTE GROUP Indication

MUTE GROUP MASTER SWITCHES

The four switches labelled 1-4 activate Mute Groups when pressed.

VCA CONTROL GROUP BEHAVIOUR

Up to 16 VCA Groups can be created.

In the normal configuration of the console with a single Output section block, there are only 8 faders available for VCA Masters, therefore Output Fader pages must be used to access all 16.

Similarly there are only 8 assignment LEDs on the Channel and Output faders, and these must be bank-switched with the display switches [VCA 1-8][VCA 9-16] in the Control Block, in order to check all the assignments if more than 8 VCAs are being used.

AUDIO BEHAVIOUR

When a VCA Group has been created, by assigning Input or Output channels to a VCA Master, the following behaviour is followed:

* The Master fader setting applies its dB value as an offset to all member channels. **The member channels' fader positions 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 fader. Any Master fader reaching –inf dB will set all member channels to –inf 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.

* The [ON]/Off switch on the VCA Master fader acts as a remote control for all member channels' On/Off switches. If a channel which was previously ON is turned OFF by the action of a VCA Master On/Off switch, the channel's ON switch 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, Green when ON. (There is no non-illuminated condition). This is because VCA Master On/Offs cannot themselves be remotely muted, so the Red helps to identify 'Muted' VCA masters.

* The [SOLO/SEL] switch on the VCA Master fader acts as a remote control for all member channels' Solo switches. The channel 'Sel' function is not activated however.

ASSIGNING VCAs

1. Press the VCA/MUTE GROUP's [SETUP] key. It will glow blue.

2. Choose the required VCA master by pressing its [SOLO/SEL] key, which will also glow blue. Note that if the output fader page is not displaying VCAs it will be necessary to select the [VCA] page (VCAs 1-8) or the [E]page (VCAs 9-16) first.

3. Press the [SOLO/SEL] key of any channel(s) that is(are) to be assigned to the selected VCA master.

4. Press [SETUP] again to finish the process, or press another VCA master's [SOLO/SEL] key to assign more channels to another VCA master.

Hint: If the SETUP mode is not switched off after assigning VCAs, channels cannot be soloed.
Hint: 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: i.e. 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.

ASSIGNING MUTE GROUPS

1. Press the VCA/MUTE GROUP's [SETUP] key. It will glow blue.

2. Press the required mute group [1-4] master key, it will glow red. The MUTE/VCA display strip on each channel will show mute assignments in red.

3. Press the [SOLO/SEL] key of any channel(s) that is(are) to be assigned to the selected MUTE master.

4. Press [SETUP] again to finish the process, or press another MUTE master's key to assign more channels to another MUTE master.

Hint: If the SETUP mode is not switched off after assigning MUTES, channels cannot be soloed.

ASSIGNING VCAS AND MUTE GROUPS TO OUTPUT CHANNELS

It is possible to assign VCAs and/or MUTE Groups to any of the 32 output channels (but not the LRC master outputs). In step 3 of either of the assignment processes above, select one of the output fader pages [A] to [D], and then use the [SOLO/SEL] key below the output fader to assign it to the VCA or Mute Group in question. Note: you must not assign a VCA group to both input and output channels.

VCA GROUPS WHEN AUX SENDS ARE CONTROLLED BY CHANNEL FADERS

The VCA Groups on the Soundcraft Vi Series™ 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. The member channels can also be Soloed or Muted, using the VCA Master [SOLO/SEL] and [ON] switches.

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

Effectively, because there can be up to 32 mono Aux 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. This means that there are up to 33 virtual sets of 16 VCA master faders in total.

VCA control of Auxes is only available by activating the Follow Output Solo [FLW] key next to the master faders, note that the pair of [FLW] keys for the VST Encoder Rows will not access this function.
Due to the Follow Output Solo functionality, it is possible to control only one Aux mix via VCAs at any one time.

Procedure

Select the Aux required by selecting the required master fader bank [A]-[D].
Activate the [FLW] key next to the master faders.

Press the [SOLO/SEL] key, under the master fader, for the required Aux. Alternatively, the Aux can be soloed using the solo switches in the Master Vistonics™ screen.

Press [VCA] (for VCA 1-8) or [E](for VCA 9-16). The FaderGlow™ for the master faders will change to blue (VCA 1-8) or pink (VCA 8-16). The faders will move to show the offsets which are being applied to the Aux feeds from those channels which are assigned to the VCAs now being displayed in the master section.

Note that 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 (the channel's [ON] switch is illuminated red if the Aux send is turned Off by a VCA Master [ON] switch.)

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].

In the example in Figure 9-3, input channel 1 has been assigned to VCA 1, and channels 2,3 & 4 have been assigned to VCA 4. In order to use the VCAs to control the Aux 5 sends from the input channels proceed as follows: select bank A, activate [FLW], solo Aux MASTER 5 using its [SOLO/SEL] key, press [VCA]. VCA 4's fader will now control the Aux 5 feeds from input channels 2, 3 & 4; and VCA 1's fader will now control the Aux 5 feed from input channel 1.

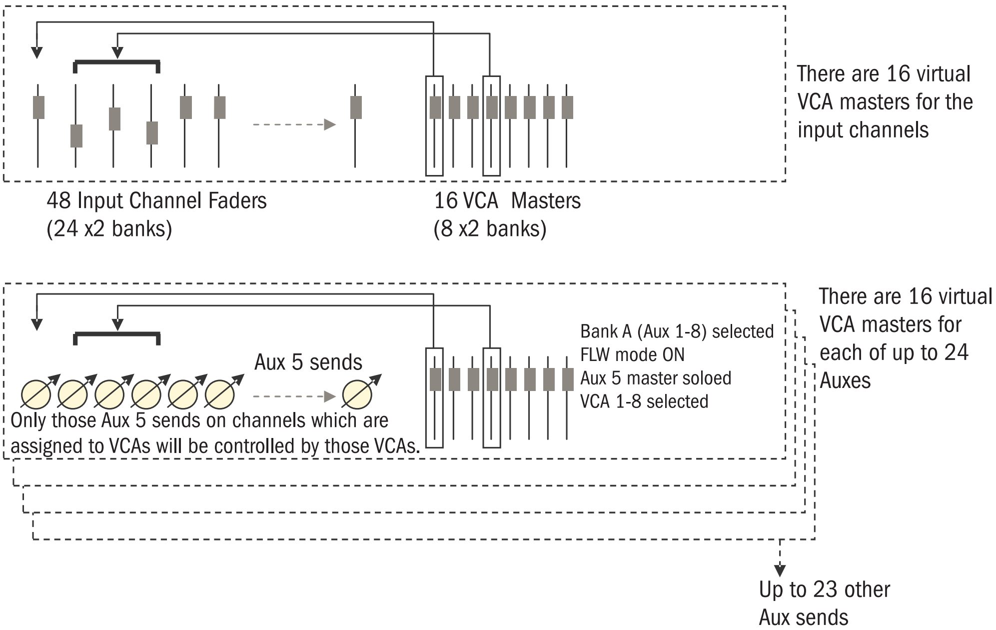


Figure 9-3. How VCA Groups Operate on Aux Sends.

SIGNAL FLOW

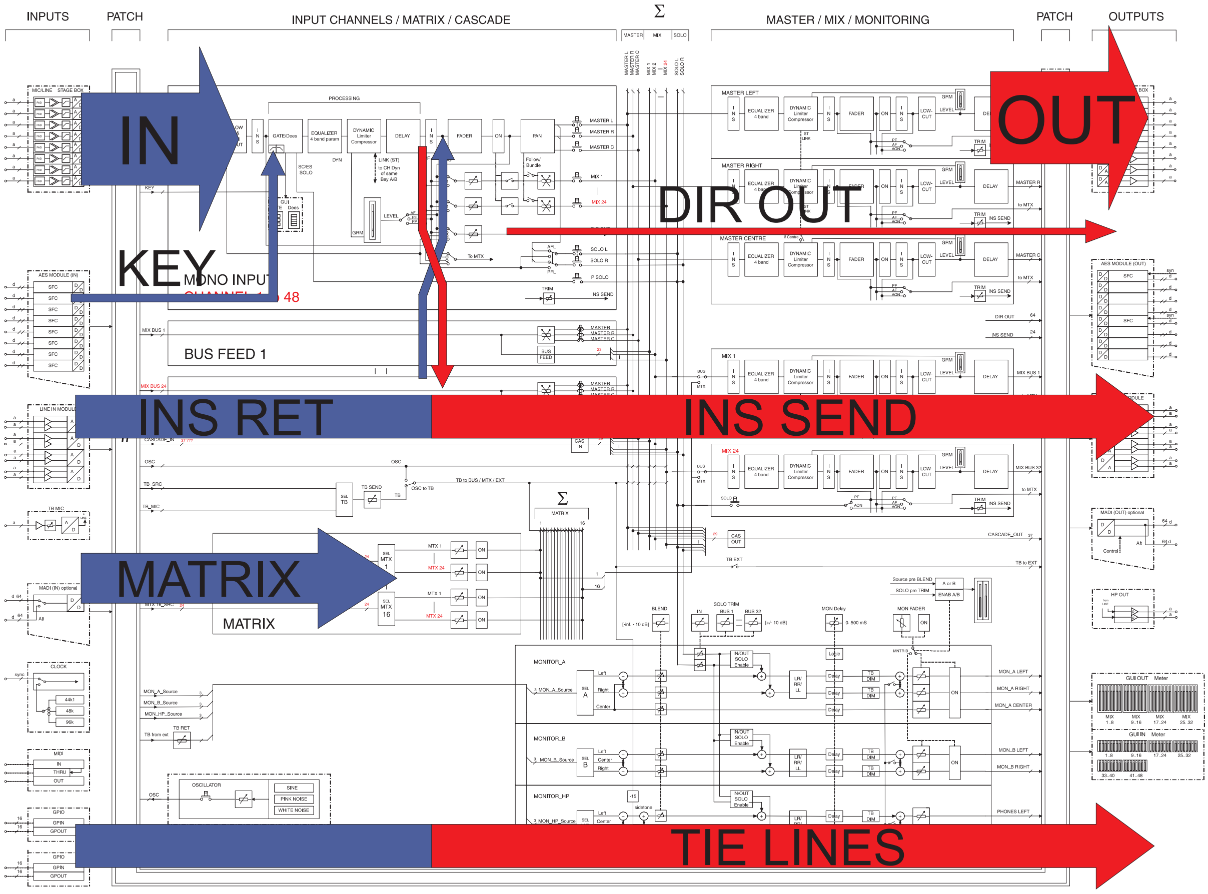


Figure 10-1: Patch Signal Flow.

OVERVIEW

The patch system of the Soundcraft Vi Series™ is divided into functional groups allowing it to be accessed easily and intuitively via the console work surface.

The following table gives an overview of where the Patch functions are located on the console work surface.

	Location			
Patch Function	**Input Bay**	**Input Bay** *[ALL BUSSES] active*	**Input Bay** *MTX [SEL] on master section active*	**Master Bay** *[SEL] active (only if [LOCK MTR] is off)*
Input	Input section			
Output		Pan section		Pan section
Insert (channel)	Pan section			
Insert (master)		Pan section		Pan section
Direct Out	Pan section			
Key Signal	Dynamics section			Dynamics section
Matrix			Direct access	
Tie Lines	[MENU] then the <TIE LINES> tab			

Figure 10-2: Patch Function Overview.

GENERAL RULES

The following rules are valid for all patch pages.

* Channel Label (name) entries are located near the Patch configuration.
* Source Patches are colour-coded BLUE, while destination patches are RED.
* Patch Pages open in the upper screen area by pressing the Patch Configuration button.
* Patch Pages close by pressing the Patch Configuration button again or by pressing the EXIT button.
* Sources and destinations are grouped location-wise (local I/O, Stage Box, Madi).
* If the Page is open, pressing SEL of an other Channel moves the Page to this bay.

INPUT

The Input Patch connects an input connector or MADI channel with the desired input channels. Each input connector signal can be patched to more than one input channel at the same time.

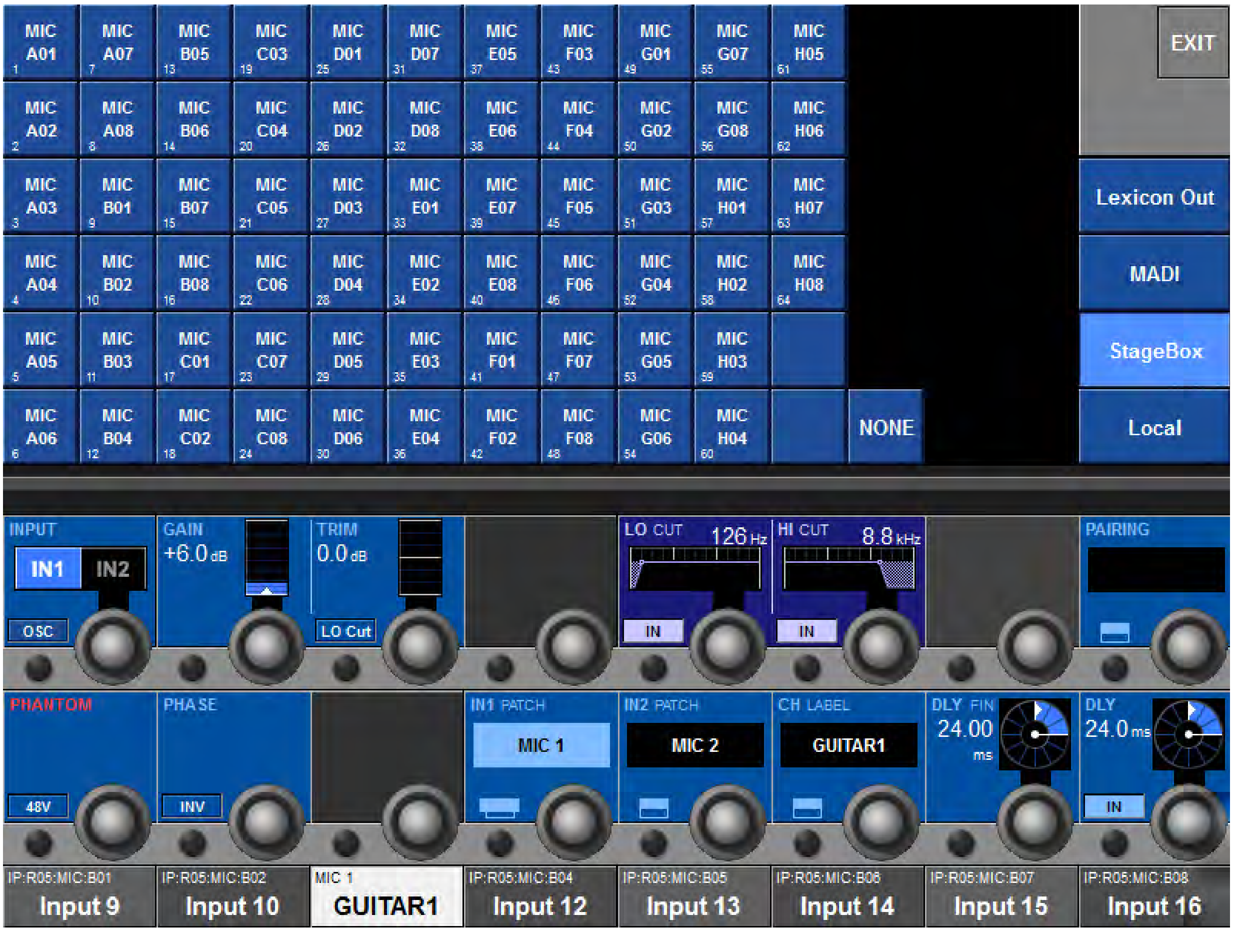


Figure 10-3: Input 1 Patch Configuration Page.

Input Patch Point can be set individually for both possible Inputs IN1 and IN2, using the IN1 PATCH and IN2 PATCH configuration button.
IN 2 can optionally be used for the > SPARE MIC function.

The small blue A/B legend in the Channel Label area shows which other channels use the same signal, where A means this channel on the Input Layer A and B means this channel on the Input Layer B.

Patch A Source To An Input Channel

* Press the Input field of the desired Channel
* Press the IN1 or IN2 Patch button
* Select the desired Input Source -> Audio will immediately patched
* Leave the Patch page by pressing the IN1 or IN2 Patch button again, or the Exit button on screen.

HINT: It is possible to select NONE, that means no audio source is patched to this INPUT.
Location groups (e.g., Stage box, Local I/O etc) can be changed by directly selecting the required group on the right-hand side of the screen.

HINT: If the channel is paired, <LEFT> and <RIGHT> comes up and allows toggling between the input patch for L & R of the paired channel.

Using A Spare Mic For Several Inputs

Figure 10-4 shows four mics patched to four input channels via each channel's IN1 patch. The spare mic is patched to all of the 4 channels via their IN2 patch. In the event that one of the main mics fails it is easy to change the appropriate channel's input from IN1 to IN2.

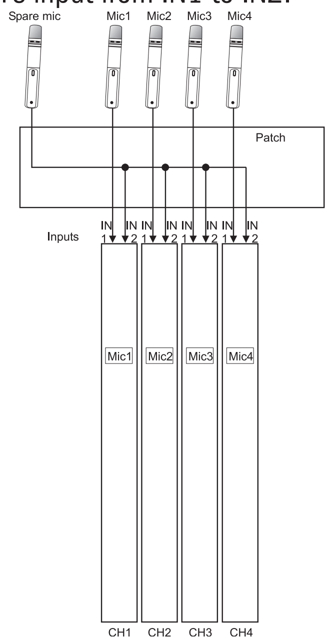


Figure 10-4: Spare Mic Connected To Several Inputs.

OUTPUT

The Output Patch connects a master or bus out with an output connector or MADI channel. A master or bus out signal can be patched to several physical outputs at the same time.

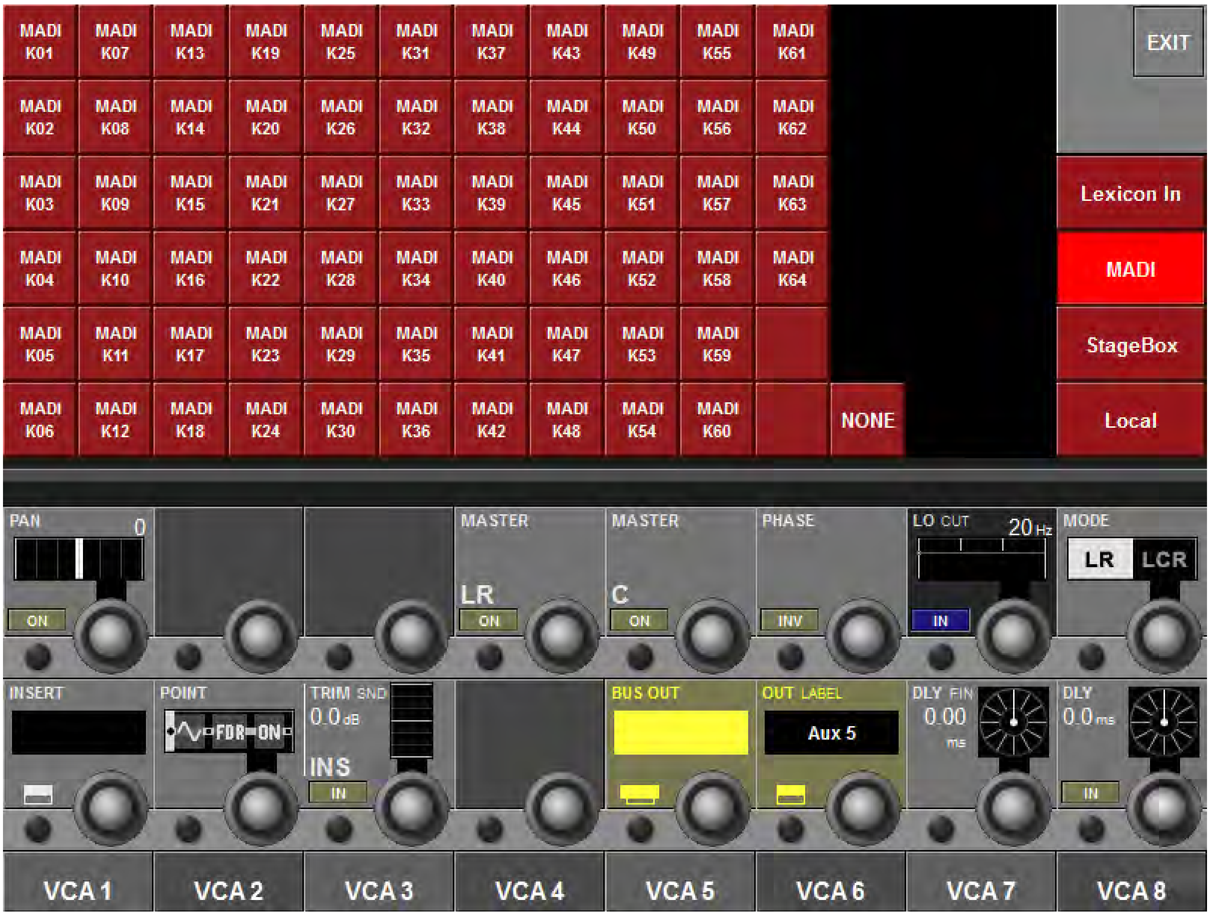


Figure 10-5: Output Patch Configuration Page.

Patch a BUS to an OUTPUT

There are two ways to access the Output Patch configuration page (Figure 10-5). They are summarised in Figure 10-2. As an example of one of the two methods proceed as follows:

* Press one of the output fader page keys [A]-[D] on the master bay.
* Ensure [LOCK MTR] is not on, and that no Setup or Menu pages are open.
* Press the [SOLO/SEL] key for the bus to be assigned to an output (or press the [SEL] key under the LR C faders to assign any of the three main output busses.
* Press the PAN area on the master section touch screen (this is in the area that the input meters were being displayed).
* Press the {BUS OUT} Patch button.
* Select the required output on the touch screen.
* Optionally select additional outputs.
* Leave the Patch page by pressing the OUT Patch button again, or the <EXIT> button on screen.

(The second method is to select the [ALL BUSSES] fader pages and access the PAN area for the required output directly on the input fader screens.)

HINT: It is possible to select <NONE> to reset the patch.

Location groups (e.g., Stage box, Local I/O, etc) can be changed by directly selecting the alternate groups.

HINT: If you adjust a STEREO Bus the desired patch page can be selected with <LEFT> and <RIGHT> .

INSERT

Figure 10-6: Insert Select Page.

Inserts are organized within an Insert POOL that contains up to 24 insert send/return pairs.. Once set up, each insert in the pool can be easily patched to the desired Input channel or master insert point.

Patching An Insert Point To An Input Channel

* Select the PAN touch field of the destination input channel.
* Press the {INSERT}VST config button. The insert select page will open in the touch area, see Figure 10-6.
* Select the preconfigured insert pair of input & output connections [1]-[24] from the pool.
* Leave the page with <EXIT> or press {INSERT} again.

Before an insert can be patched into a channel, the physical connectors or MADI channels for the send and return must be defined. A specific pair of connectors can be set up for each device, and the device name entered for easy recognition. Press <SETUP> to access the Insert Point Setup page (see Figure 10-7).

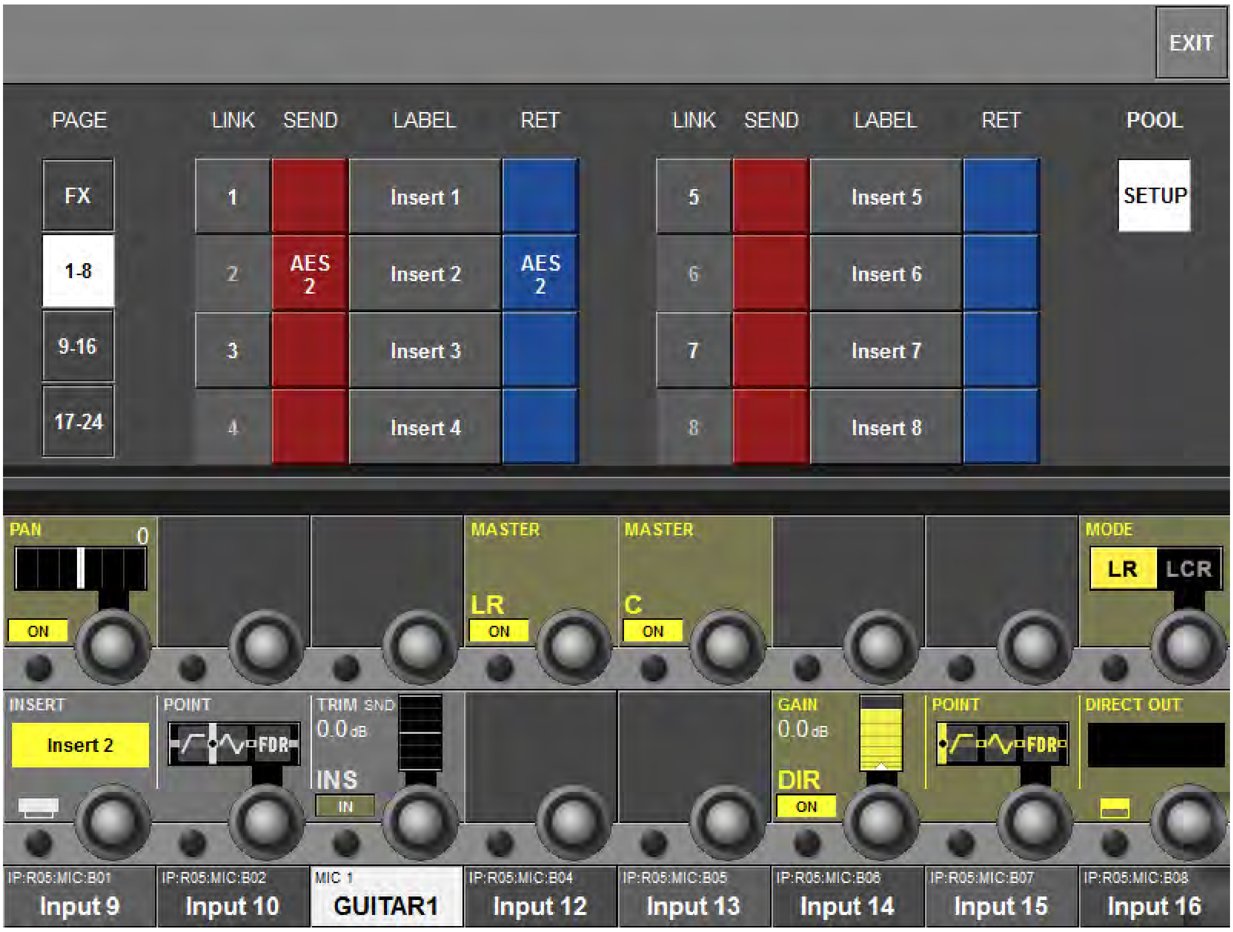


Figure 10-7: Insert Setup Page.

Patching Insert Send Or Return Signals To The Connectors Or MADI Channels

* <SEND> opens the output patch page, where it is possible to define the physical connector for the insert send.
* <RET> opens the input patch page, where it is possible to define the physical connector for the insert return.
* <LABEL> opens the keyboard page, where it is possible to label the insert.
* Leave the page with <EXIT> or press the INSERT {VST config button} again.

Stereo Inserts

Stereo inserts can be configured odd/even wise with <LINK> from the even insert number.
The following table shows the valid Format combinations.

	MONO CHANNEL	**PAIRED** CHANNEL or STEREO BUS
MONO INSERT	√	X
STEREO INSERT	√ **audio is feed to both insert sends, returns are down mixed**	√

DIRECT OUT

The direct out patch connects a channel direct out with an output connector. Direct out can be patched to several outputs at the same time.

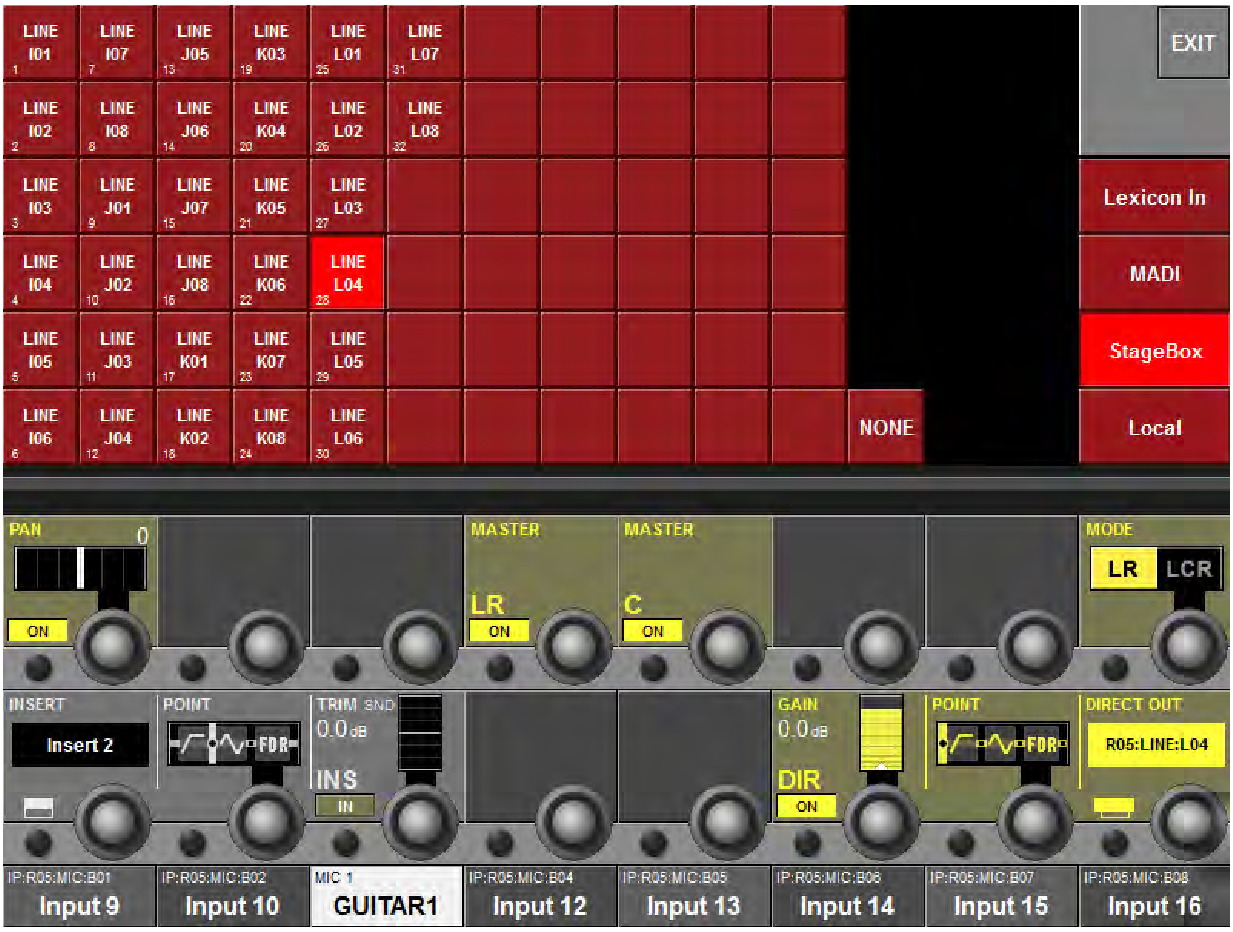


Figure 10-8: Direct OUT Patch Configuration Page.

HINT: If the channel is paired, <LEFT> and <RIGHT> comes up and allows toggling between the L & R direct out patch of the paired channel.

KEY SIGNAL

The key signal patch feeds a channel direct out or an input signal from the patch to the key input of the GATE.

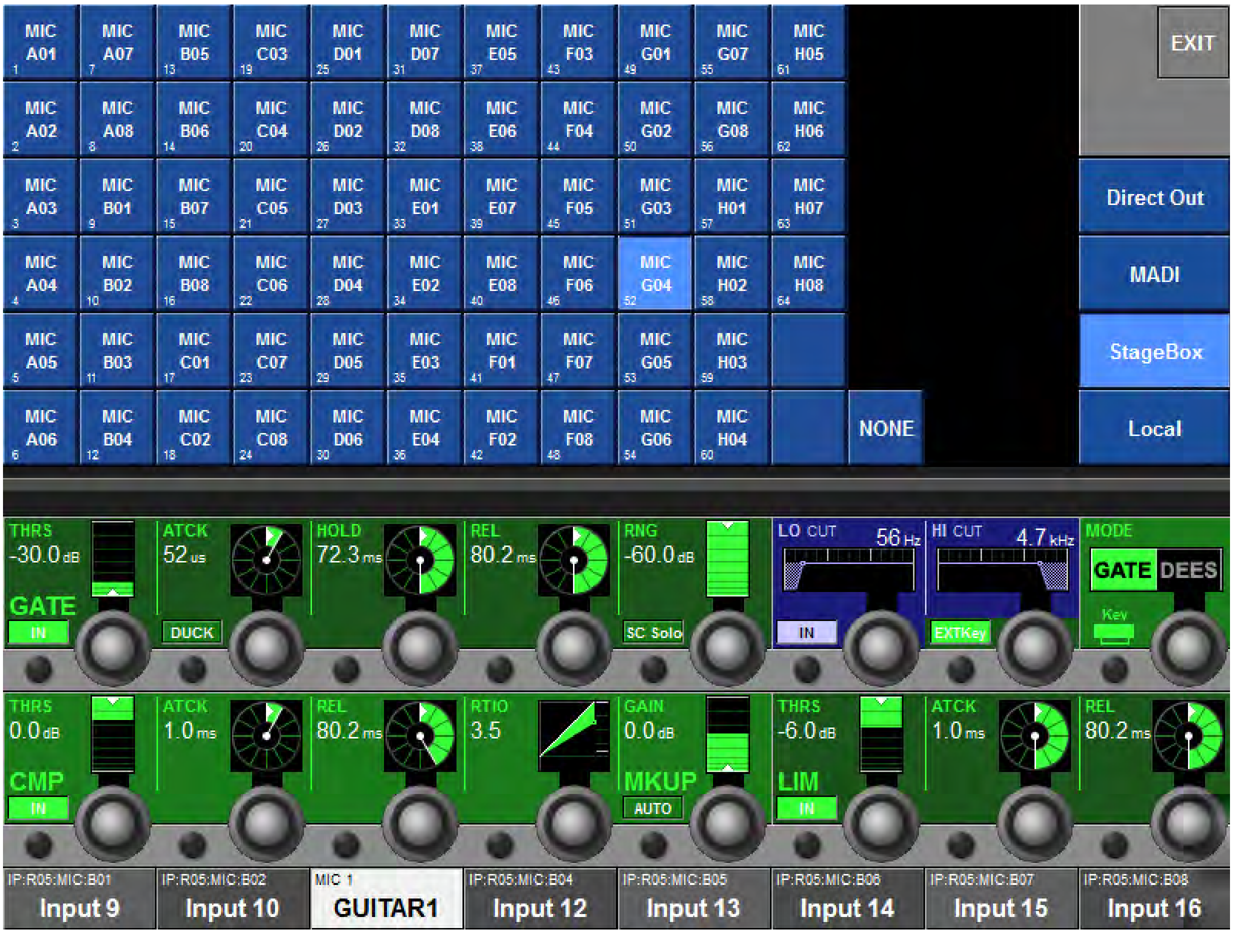


Figure 10-9: Key Signal Patch Configuration Page.

MATRIX

The matrix patch page connects a channel direct out, an input signal, MADI channel, or a master signal to become a source for a Matrix output.

Figure 10-10: Matrix Source Patch Page.

Pressing the required VST key opens the Matrix Source Patch Page.

TIE LINES

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 so do not use up any DSP channels.

To open the Tie Line Setup page press the [MENU] key, and then press the <Tie Lines> tab at the top of the master area touch screen

Figure 10-11: Tie Line Setup Page.

Soundcraft Vi Series™ supports up to 24 tie lines. 8 of them are arranged per page. The desired page can chosen using the <1-8>, <9-16>,<17-24> buttons..
<IN> opens the input patch configuration page, while <OUT> opens the output patch configuration page.

Example: Send An Audio Signal From The Stage To The FOH Location

To set up a TIE line:

* Patch the stage box input connector to a free tie line input <IN> (blue).
* Patch the TIE line to an output connector on the Local Rack <OUT> (red).
* Optionally re-label the tie line <LABEL>.

SIGNAL FLOW

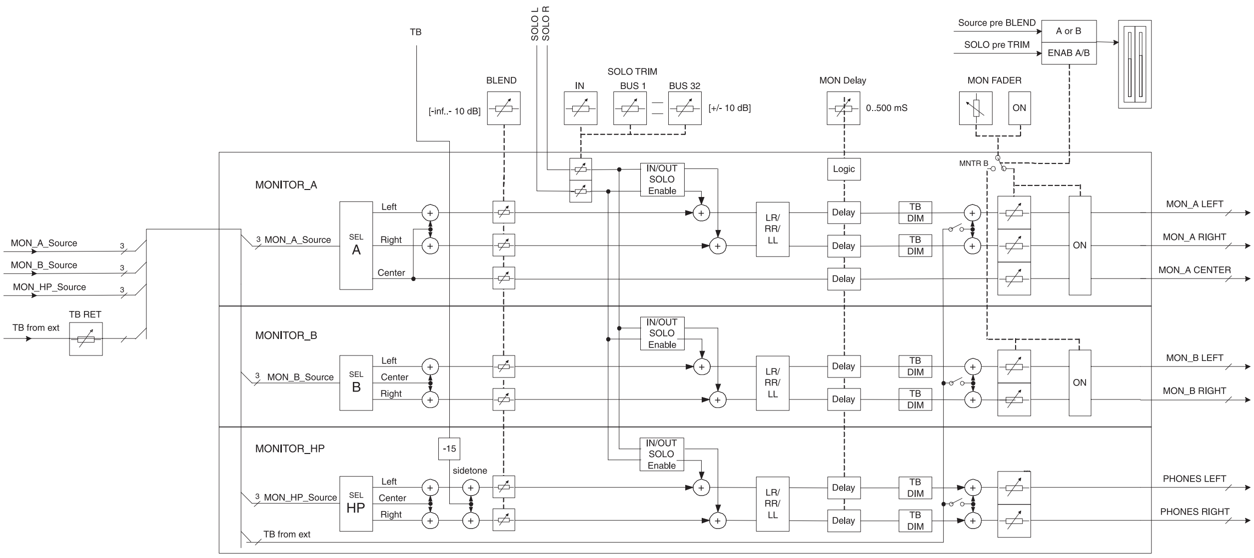


Figure 11-1: Monitor Block Diagram.

FUNCTION

The Monitoring section in the Soundcraft Vi Series™ has three individual monitoring outputs:

Monitor A	3 channel (LCR) format (it can be used as stereo by ignoring C)
Monitor B	Stereo format
Headphones	Stereo format.

For each monitoring output, the following parameters can be set or configured:

- Source
- Input SOLO
- Output SOLO (with user configurable OUT SOLO Group)
- TB from external
- Audio Format (swapping or mono-ing Left or Right channels).

Two alternative monitoring sources, USER A and USER B, can be freely assigned and labelled. These can be used for a favourite monitor mix, or a ‘shout’ talkback feed.

DESK VIEW

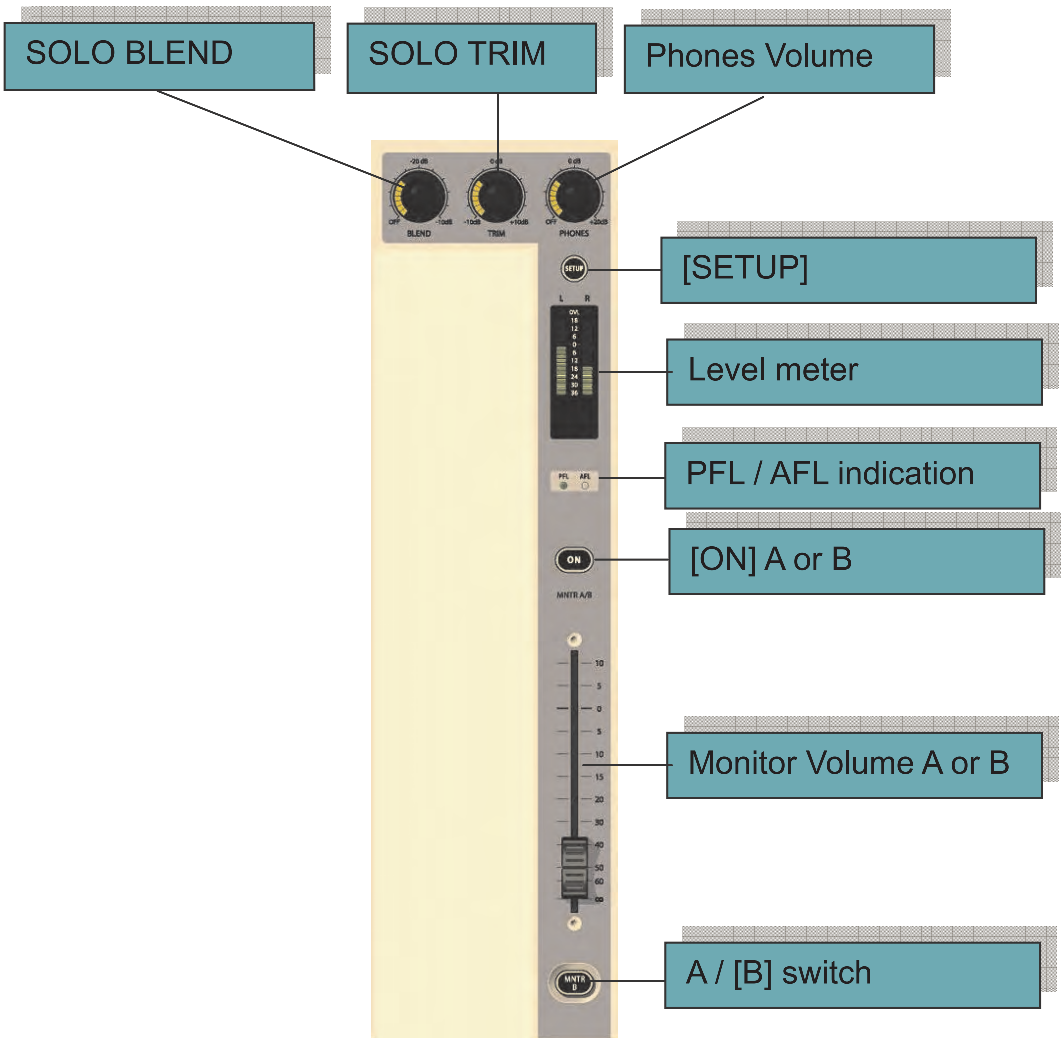


Figure 11-2: Monitor Controls.

SOLO TRIM
Trims the SOLO Level in the range +/- 10 dB. 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.

SOLO BLEND
Adjusts the background level of the monitoring source signal which is audible when a solo is in operation from OFF (-∞)(as normal consoles) to a –10dB dim. This allows solos to be heard within a mix which has a reduced background level.

PHONES Volume
This encoder is permanently assigned to 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.

SETUP
When this is pressed the Monitor Setup Page (see Figure 11-3) is displayed on the master screen. The setup key glows blue when it is active.

Level Meter
The stereo Level Meter shows the level of the A or B outputs, depending upon the selection made via the Monitor B switch. The meters follow the Monitor Volume Fader.

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

ON
This switches the currently-selected monitor (MTR A or MTR B) on.

Monitor Volume
This controls the volume of the currently-selected monitor (MTR A or MTR B).

Monitor B
This selects 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.

Hint: both monitor A and B continue to operate irrespective of the selection made by the Monitor B switch.

MONITOR SETUP PAGE

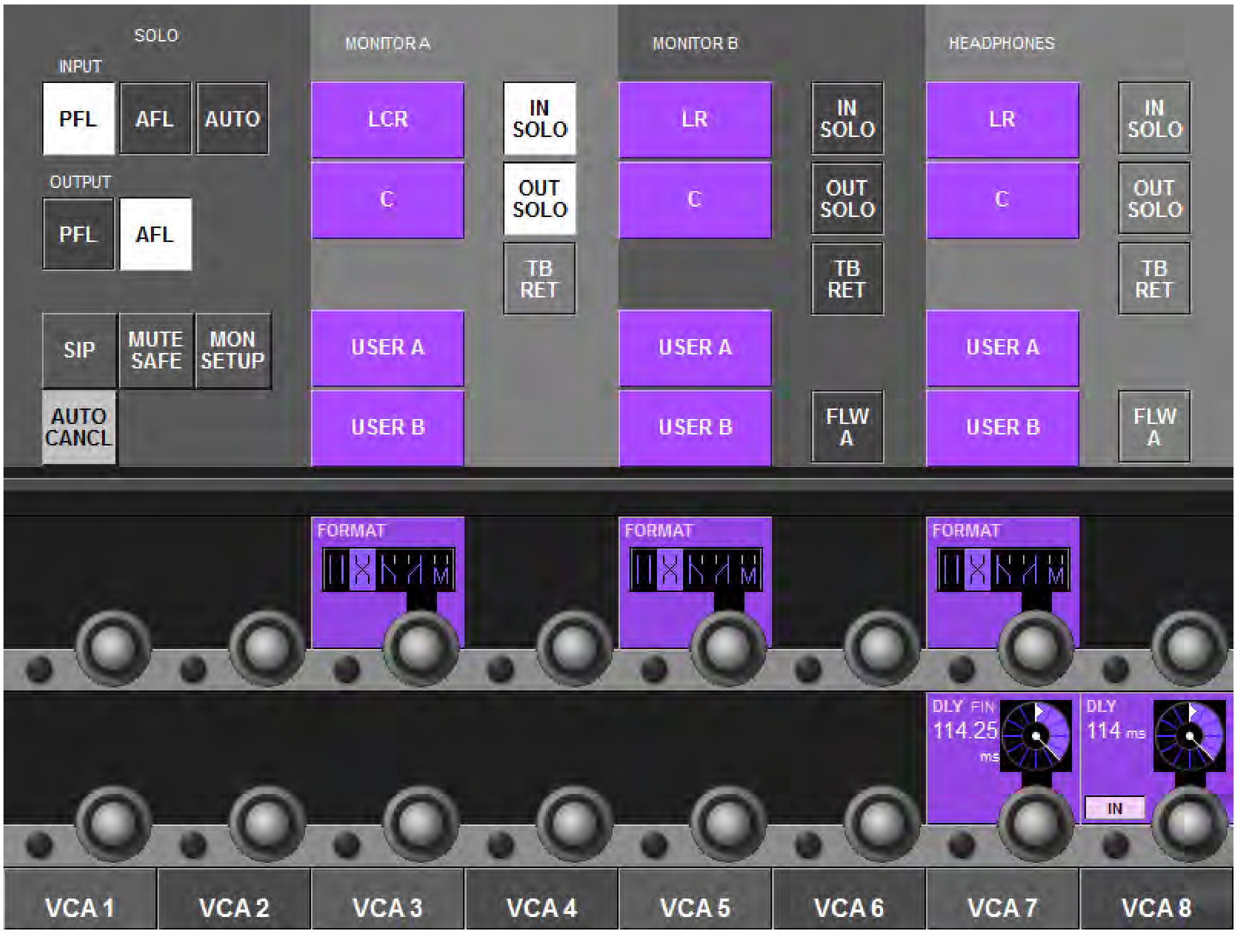


Figure 11-3: Monitor Setup Page.

SOLO Section

Input

<PFL>

Sets the input channel solo mode to PFL.

<AFL>

Sets the input channel solo mode to AFL.

<AUTO> (default)

Sets the input channel solo mode automatically, as follows. If 1 Input SOLO is active the mode is PFL, if more then 1 Input SOLO is active at the same time the mode is AFL. (Press and hold the first input solo key to then select additional solos.)

Output

<PFL>

Sets the Output SOLO mode to PFL.

<AFL> (default)

Sets the Output SOLO mode to AFL.

Miscellaneous

<SIP>

Activates the SOLO-IN-PLACE mode. This is a destructive mode for use only during soundchecks 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>
Enables 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. This configuration mode is disabled when the setup page is exited.
<MON SETUP>
Enters the monitor setup sub-page (see Figure 11-6).

MNTR A Section

	Mon A Source Options							Mon A Audio Out		
	LCR	**C**	**USER A**	**USER B**	**IN SOLO**	**OUT SOLO**	**TB RET**	**L**	**C**	R
Normal Monitoring (No SOLO active)	✔							L	C	R
	✔	✔						L+ (C-3dB)	C	R+ (C-3dB)
		✔				don't care		C		C
			✔					USR A L		USR A R
				✔				USR B L		USR B R
Input SOLO is active					✔	don't care	don't care	The Input channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.		
	don't care					don't care	don't care	as selected by LCR/C/USR A/USR B		
Output SOLO is active	Note that if the SOLO BLEND control is not at $-\infty$ a proportion of the normally-monitored signal will be heard during a Solo activation.				don't care	✔	don't care	The Output channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.		
					don't care		don't care	as selected by LCR/C/USR A/USR B		
TB Return is active					don't care	don't care	✔	The TB Return channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.		
					don't care	don't care		as selected by LCR/C/USR A/USR B		

Figure 11-4: Summary Of Monitor A Option Functionality.

Source
USER A, USER B and (LCR,C) are mutually exclusive, but LCR and C can be mixed. Also none can be selected.
<LCR>
Sets the monitor A Source to LCR.
<C>
Sets the Monitor A Source to C.
<USER A>
Sets the Monitor A Source to USER A. This could, for example, be used for a 2-track return.
<USER B>
Sets the monitor A Source to USER B. This could, for example, be used for a 2-track return.

Solo Switching

<IN SOLO>

Routes the Input SOLO Signals to the Monitor A Output.

<OUT SOLO>

Routes the Output SOLO from the Busses that are included in the OUT SOLO selection (see Figure 11-6) to the Monitor A Output.

<TB RET>

Routes the Talkback Return Signal to the Monitor A output.

Format field

The encoder selects one of the following options to be the listening format (the icons are shown to the left of the list):

LR	left source to left monitor, right source to right monitor
RL	left source to right monitor, right source to left monitor
LL	left source to left and right monitor
RR	right source to left and right monitor
Mono	left and right source is summed and fed to left and right monitor. The Centre signal is not affected.

MNTR B and HP Sections

	Mon B/H'phones Source Options							Mon B/HP Audio Out	
	LR	C	USER A	USER B	IN SOLO	OUT SOLO	TB RET	L	R
Normal Monitoring (No SOLO active)	✔					don't care		L	R
	✔	✔						L+ (C-3dB)	R+ (C-3dB)
		✔						C	C
			✔					USR A L	USR A R
				✔				USR B L	USR B R
Input SOLO is active	don't care				✔	don't care	don't care	The Input channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.	
						don't care	don't care	as selected by LR/C/USR A/USR B	
Output SOLO is active	Note that if the SOLO BLEND control is not at -∞ a proportion of the normally-monitored signal will be heard during a Solo activation.				don't care	✔	don't care	The Output channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.	
					don't care		don't care	as selected by LR/C/USR A/USR B	
TB Return is active					don't care	don't care	✔	The TB Return channel's signal is routed to the monitor, the LCR busses are configured by its status as a mono or stereo-paired channel.	
					don't care	don't care		as selected by LR/C/USR A/USR B	

Figure 11-5: Summary Of Monitor B and Headphones Option Functionality.

Source
USER A, USER B and (LR,C) are mutually exclusive, but LR and C can be mixed. Also 'NONE' can be selected.
<LCR>
Sets the monitor A Source to LR.
<C>
Sets the Monitor A Source to C.
<USER A>
Sets the Monitor A Source to USER A.
<USER B>
Sets the monitor A Source to USER B.

Solo Switching
The following four fields apply to the MTR B and Headphones sections of the screen page
<IN SOLO>
Routes the Input SOLO Signals to the Monitor B/Headphones Output, according to which of the two possible fields are selected.
<OUT SOLO>
Routes the Output SOLO from the Busses that are included in the OUT SOLO selection (see Figure 11-6) to the Monitor B/Headphones Output, according to which of the two possible fields are selected.
<TB RET>
Routes the Talk Back Return Signal to the Monitor B/Headphones Output, according to which of the two possible fields are selected.
<FLW A>
Selecting this option forces the source selection for Monitor B and/or Headphones to follow the selection made for Monitor A. All the other options for Monitor B and/or Headphones are disabled when FLW A is enabled.

Format Field
Selects the listening format.
The encoder selects one of the following options to be the listening format:

LR	left source to left monitor, right source to right monitor
RL	left source to right monitor, right source to left monitor
LL	left source to left and right monitor
RR	right source to left and right monitor
Mono	left and right source is summed and fed to left and right monitor.

DLY Field

The encoder changes the monitoring delay in the range 0 - 2000 mS. in several ms steps. A second control allows fine adjustment in 0.02ms steps.

{IN} enables the delay function.

This parameter applies to all three monitor circuits. 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.

MONITOR SETUP SUB-PAGE

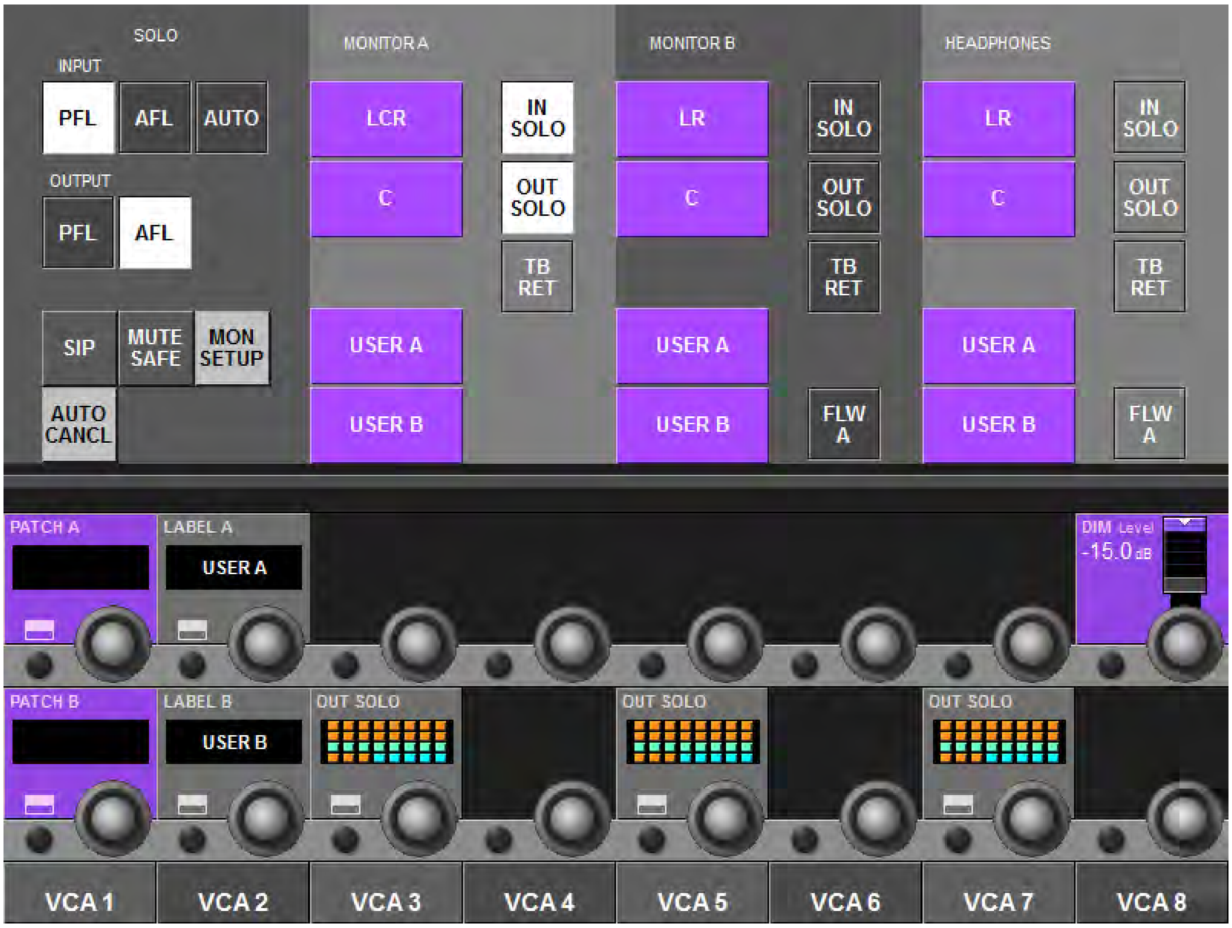


Figure 11-6: Monitor Setup Sub-page.

PATCH A field
Displays the source name that is patched to USER A. Its {VST config button} opens the USER A patch page (see Figure 11-7).

LABEL A field
Displays the USER A label. Its {VST config button} opens the USER A label configuration page, which displays the internal keyboard, and allows the USER A label to be edited.

DIM LEVEL field
The encoder adjusts the DIM Level between 0 and – infinity. The DIM function is only activated if the Return Talkback function is activated via the GPIO facility (see Chapter 16).

PATCH B field
Displays the source name that is patched to USER B. Its {VST config button} opens the USER B patch page, which is similar to Figure 11-7.

LABEL B field
Displays the USER B label. Its {VST config button} opens the USER B label configuration page, which displays the internal keyboard, and allows the USER B label to be edited.

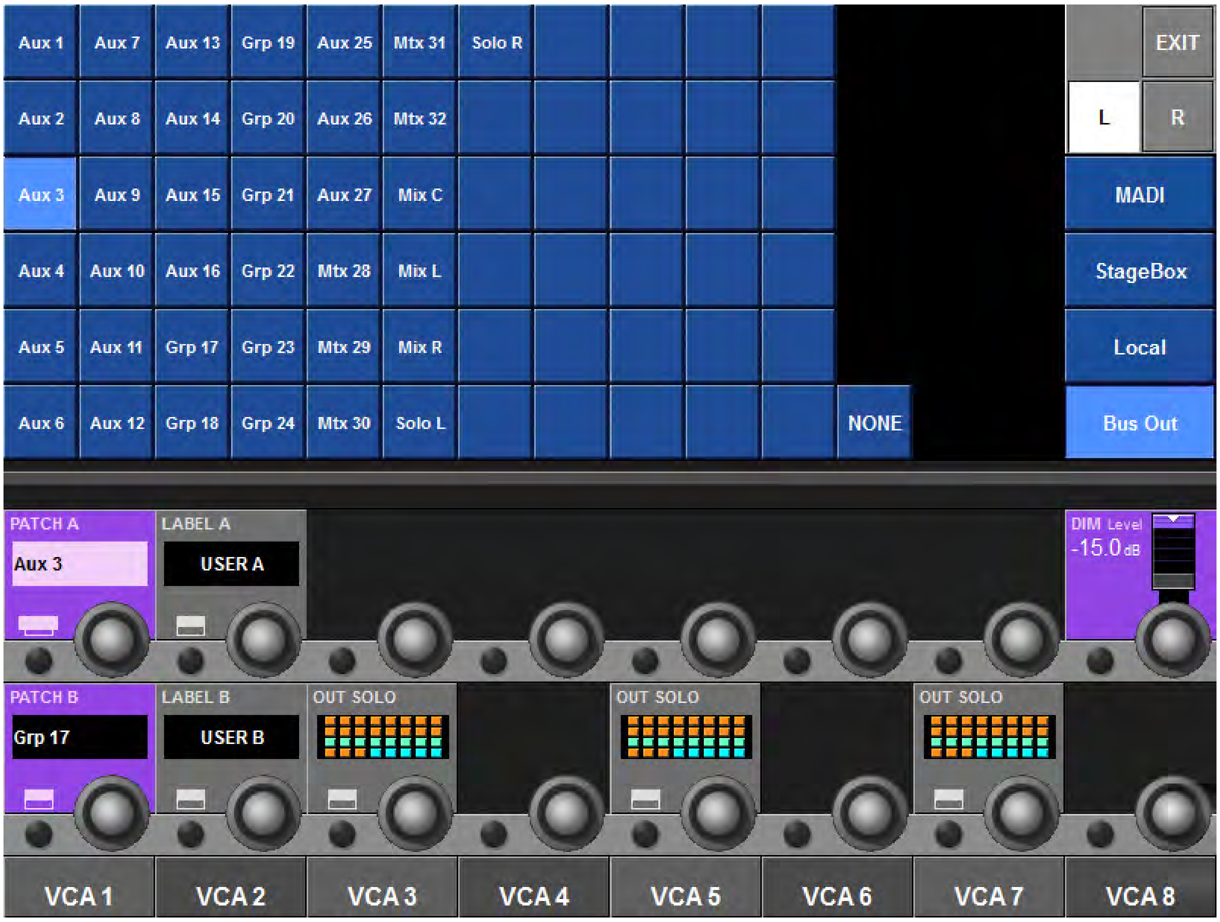


Figure 11-7: User A Patch Page

The USER A and USER B patch pages allow an alternative monitor source to the normal LR or LCR mix, to be set up.

This can be used, for example, to allow the user to listen to a favourite monitor mix, whenever a solo is not pressed (select an Aux bus from the Bus Out page, as shown above).

Another application is to allow an input from the Stagebox to be used as a 'shout' talkback line: if the relevant Stagebox input is selected as the User A or B source, this can be monitored whenever no desk solos are being monitored.

The User A and User B sources are normally stereo, but a mono source can be used by patching the same source to L and R channels in the patch page shown above.

Output Solo Selection

Output Solo Selection allows an individual output solo to be sent only to a specific monitor output (Mon A, B or Phones). 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.

OUT SOLO A field

Displays the Busses which are included in the OUT SOLO selection (default ALL) for Monitor A. Its VST {config button} opens the OUT SOLO Group configuration page (Figure 11-8).

OUT SOLO B field

Displays the Busses which are included in the OUT SOLO selection (default ALL) for Monitor B. Its {VST config button} opens the OUT SOLO Group configuration page (Figure 11-8).

OUT SOLO HP field

Displays the Busses which are included in the OUT SOLO selection (default ALL) for the headphones. Its {VST config button} opens the OUT SOLO selection configuration page (Figure 11-8).

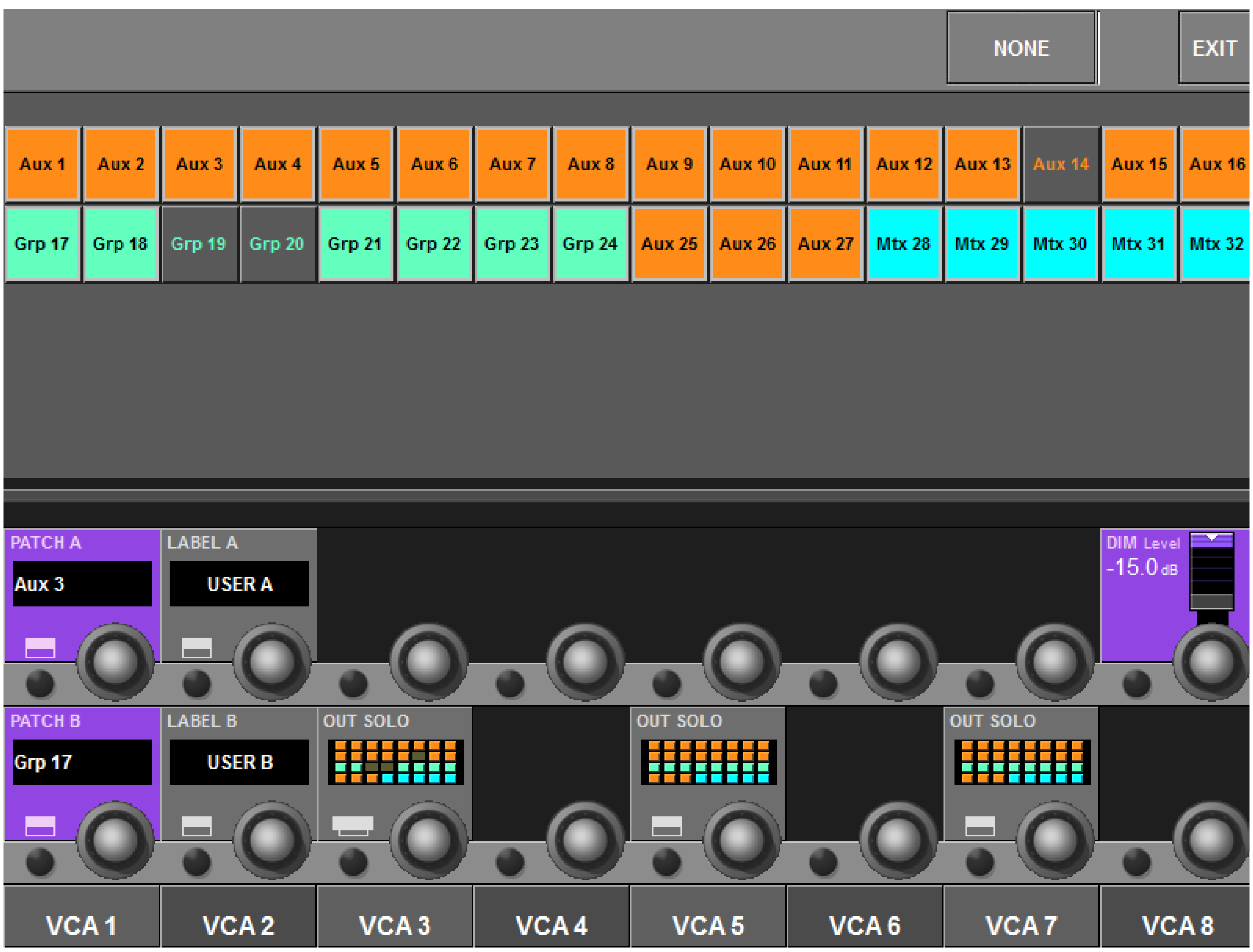


Figure 11-8: Output Solo Selection Page

Each of the 32 outputs can be switched in or out of the output monitoring group, for each of the 3 monitoring circuits. The <NONE> key deselects all of them. <EXIT> makes the display go back to the previous page. The selctions made are mirrored in the small OUT SOLO areas of the VST area. The colours indicate the type of each output: Aux Group or Matrix.

SOLO SYSTEM

AFL ,PFL and SIP

The solo system on Soundcraft Vi Series™ comprises a non-destructive PFL and AFL Solo capability from Inputs and Outputs, and also the option of a destructive Solo-In-Place mode, for use only during soundchecks 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.

If Solo-In-Place (SIP) Mode is OFF

* Soloing a single Input locally generates a PFL Solo onto the Solo Bus

* Soloing an Input (or group of Inputs) by soloing a VCA Master that the input is assigned to, generates an AFL Solo onto the Solo Bus (even if the Input Solo Mode is set to PFL in the Monitor Setup page).

* 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) will generate an AFL Solo onto the Solo Bus from all soloed Inputs.

* The PFL feed from Mono Inputs is independent of channel Pan, and is fed to the stereo Solo Bus as centre-panned image. 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 Inputs is stereo and follows the channel Pan. If the Input is a Stereo Input the AFL feed is stereo and follows the channel Balance control.

* Soloing an Output (either locally or via a VCA Master Solo) generates an AFL Solo. The post-fade Output signal is switched onto the Solo Bus.
If the Output is a Mono Aux, Group or Matrix, the signal is fed to both left and right Solo Bus equally (ie centre-panned image). If the Output is linked as a stereo pair, the signal from left and right Outputs are fed to left and right Solo bus respectively. There is no manually controlled Pan on Output Solos.

* 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.

If Solo-In-Place (SIP) mode is ON

* Soloing an Input generates a ‘destructive’ SIP Solo, muting all other Inputs which are not Soloed or set to Mute Safe. Other channels which are subsequently soloed will be unmuted.

* 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.

* Soloing an Output generates a normal Output AFL Solo, 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 Solo audio is heard on the Monitors, replacing the previous monitor source, if any.

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 Config 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 (see Figure 11-9). This switch illuminates when any Solos are active.

* Solos can be switched off manually.

* Solos can be cleared by pressing other Solos, under the rules of the Autocancel system (see later in this chapter).

Figure 11-9: SOLO CLEAR Key

Input Priority Mode

When I/P Priority is enabled, via the Input Priority key (see Figure 11-10), 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.

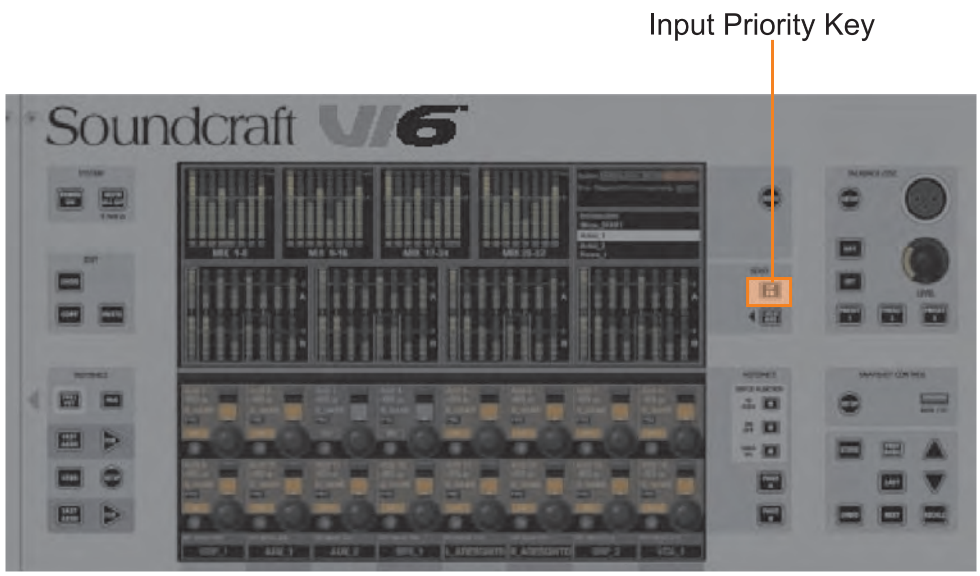


Figure 11-10: INPUT PRIORITY Key

Autocancel Behaviour

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. The Soundcraft Vi Series™ includes an optimised version of this system, allowing solos to autocancel in normal operation, but also allowing the operator to select multiple solos at once if required.

Input Priority OFF

* Pressing any single Input or Output Solo will cancel any other active Solo of either type.

* If an Input or Output Solo is pressed and held (whether already active or not), then one or more other Solos are also pressed, the autocancel behaviour is bypassed, and multiple Solos can be selected. Input Solos can change from PFL to AFL in this case if the AUTO mode is selected for input solos, in the Monitor Setup page. Pressing any Solo after the first Solo is released will cancel all the active Solos.

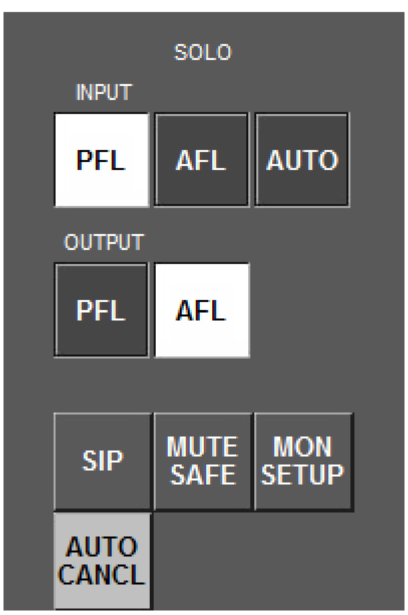
Input Priority ON

* Pressing any single Input Solo will cancel any other active Input or VCA Solo(s), and will temporarily override (but not cancel) any active Output Solo(s), as described above.

* Pressing any single Output Solo will cancel any other active Output Solo(s).

* The autocancelling can be defeated by holding down an Input or Output Solo and then pressing other solos of the same type. (Pressing the 'other' type of solo in this condition is ignored).

Switcahble AutoCancel Mode



The AUTO CANCL button allows an operator to choose whether to work with additive or auto-cancelling Solo switches.

When AutoCancel is switched off, a specific channel can be left in Solo mode to act as a return talkback feed, whilst other channels are also soloed one by one during a line check.

Follow Output Solo Mode

Purpose Of The Follow Output Solo Mode

The Follow Output Solo [FLW] keys allow the user to quickly identify and adjust those input channels which are making contributions to each of the 32 outputs.

There are three [FLW] keys: there is one fader key and two Vistonics™ area keys. Only one key can be active at any one time (none can be selected also). Their location is shown in Figure 11-11.

* 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 Vistonics Output screen (in the space normally occupied by the Input Meter display) unless the 'Lock Meters' switch adjacent to the Vistonics display has been pressed.

* 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.

* 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.

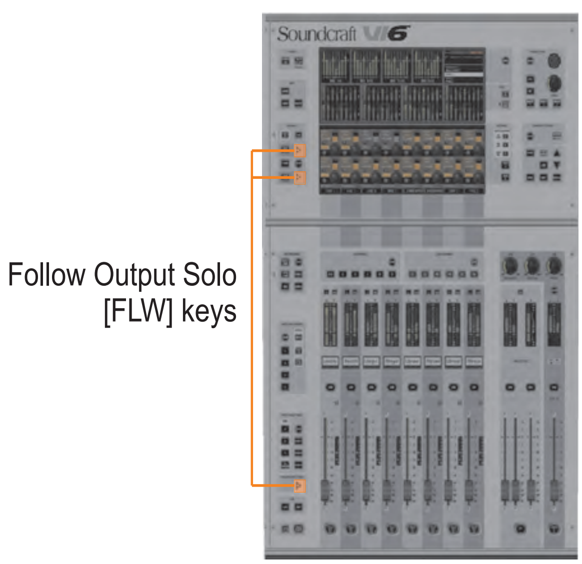


Figure 11-11: Location Of FOLLOW SOLO keys.

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

DESK VIEW

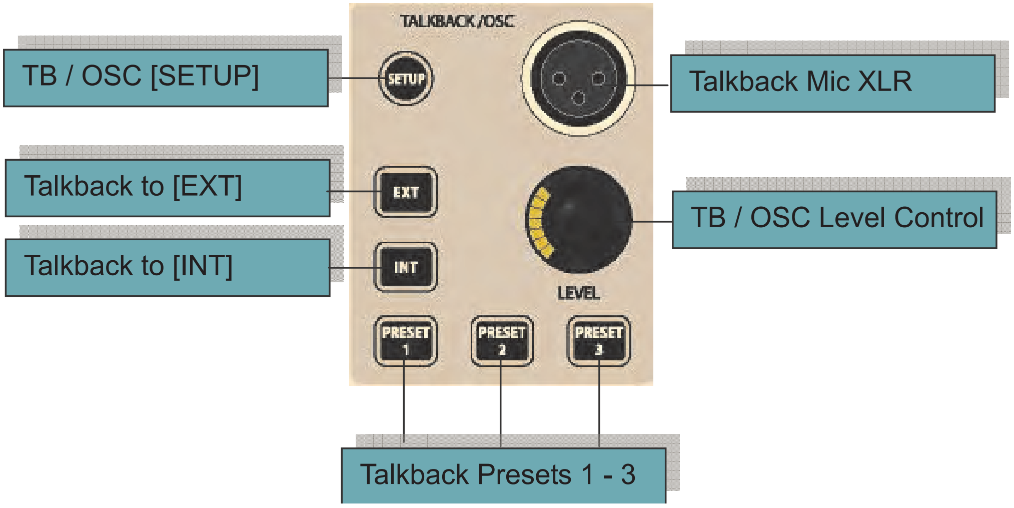


Figure 12-1: TB/Oscillator Panel Controls.

Setup Key

The [SETUP] key opens and closes a dedicated Config page on the Output Vistonics™, allowing control over the following Talkback and Oscillator functionality: Mic gain and routing destinations for the three preset buttons, routing of return TB to busses and/or phones; Oscillator waveform, freq, mode and routing.

The remaining controls on the front panel are for the talkback system only.

TB Mic XLR

There are 2 parallel-connected mic inputs: one on the front panel of the console, and one on the rear of the console. The switch for phantom power for both sockets, if it is required, is next to the rear panel socket.

TB /Osc Level control

This front panel encoder gives real-time control of TB or Osc level. For Talkback, the analogue Mic Amp gain is set at one of three values (46dB, 56dB or 66dB) by an internal jumper which is located on the pcb that holds the rear panel TB mic socket. If the Oscillator is active, the control adjusts the Osc level to all destinations. The level setting is stored independently for TB and Osc.

Routing the TB signal

The following keys route the internal TB mic (or other TB source - see Figure 12-5) signal.

INT

The TB signal is routed to preselected output busses. Output busses are selected via their {TB} VST keys in the VST area of the master screen. The {TB} keys are enabled by selecting the [TB ASSN] key which is located to the right of the master screen's VST area. The output busses are displayed in two ranges, 1-16 and 17-32, which are controlled via the [PAGE A] and [PAGE B] keys (see Figure 7-6).

EXT

The TB signal is routed to an external balanced analogue line output on the Local Rack (TB OUT), or to a choice of outputs on the Local Rack, Stage Box or MADI interface (see Figure 12-6).

PRESET 1 - 3

Presets 1-3 are user-programmable press-to-talk (momentary & latching) switches which talk directly to specified outputs.

SETUP

[SETUP] opens the configuration page which contains the central oscillator, Talkback Send and Talkback return configuration of the Soundcraft Vi Series™

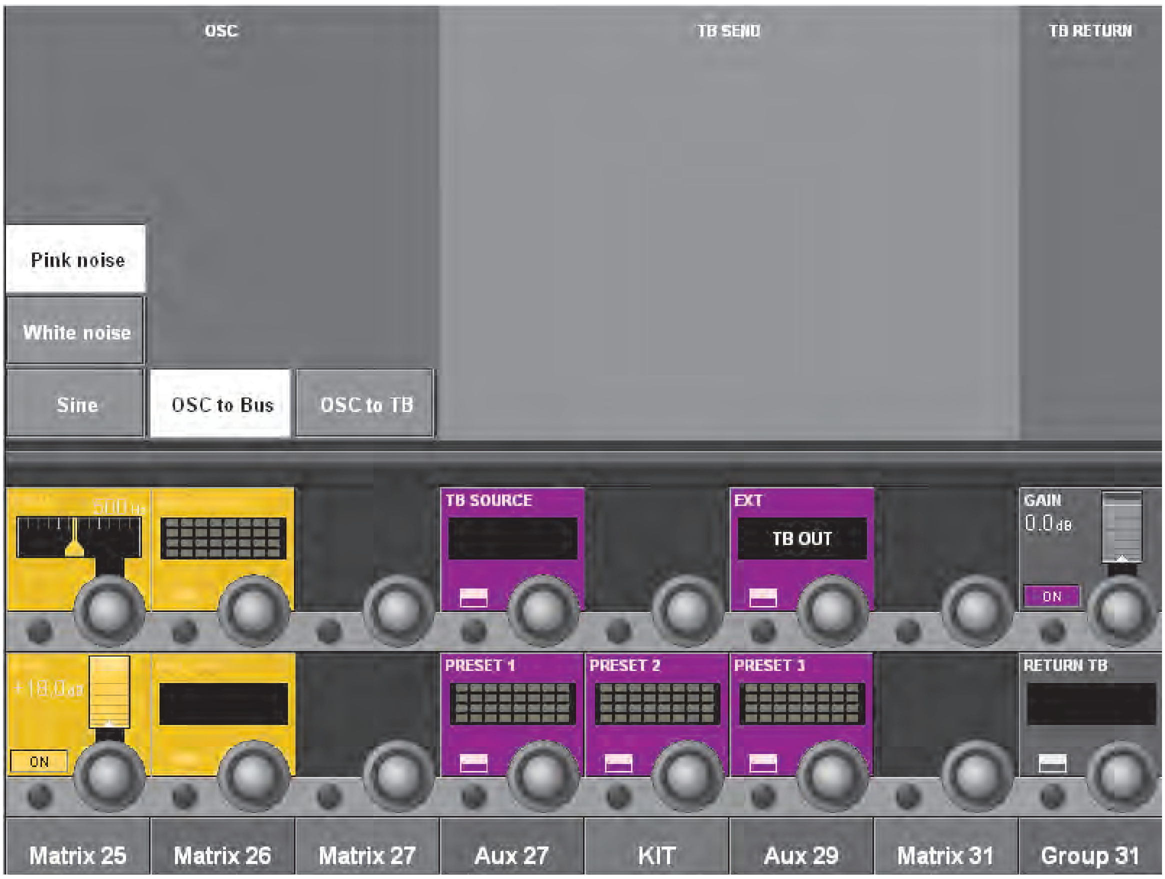


Figure 12-2: Setup Page.

OSC Section

HINT: The oscillator can also be patched to the Input Channels by using the {OSC} key in the Input Channel VST screen.

When the oscillator state is highlighted in the input touch field. The highlight appears whenever the OSC button in the channel's input section is enabled, regardless of whether the Oscillator is actually switched on in the TB/Osc master section

<PINK NOISE>
Sets the waveform to pink noise.

<WHITE NOISE>
Sets the waveform to white noise.

<SINE>
Sets the waveform to sine.

Figure 12-3: Bus Assign Page.

<OSC to BUS>

Feeds the Oscillator to the preconfigured Busses, which are selected in the Bus Assign page (Figure 12-3).

<OSC to TB>

The oscillator signal is routed to the TB Bus and replaces the TB signal.

FREQ field

If the oscillator is set to SINE the Encoder adjusts the Frequency in the range 20 Hz – 20kHz.

BUS ASSIGN field

Its {VST config button} opens the Buss Assign page (see Figure 12-3).

GAIN field

Encoder adjusts the oscillator level in the Range – inf to +12 dB. {ON} enables the oscillator.

OSC OUT field

Its {VST config button} opens the output patch configuration page (see Figure 12-4).

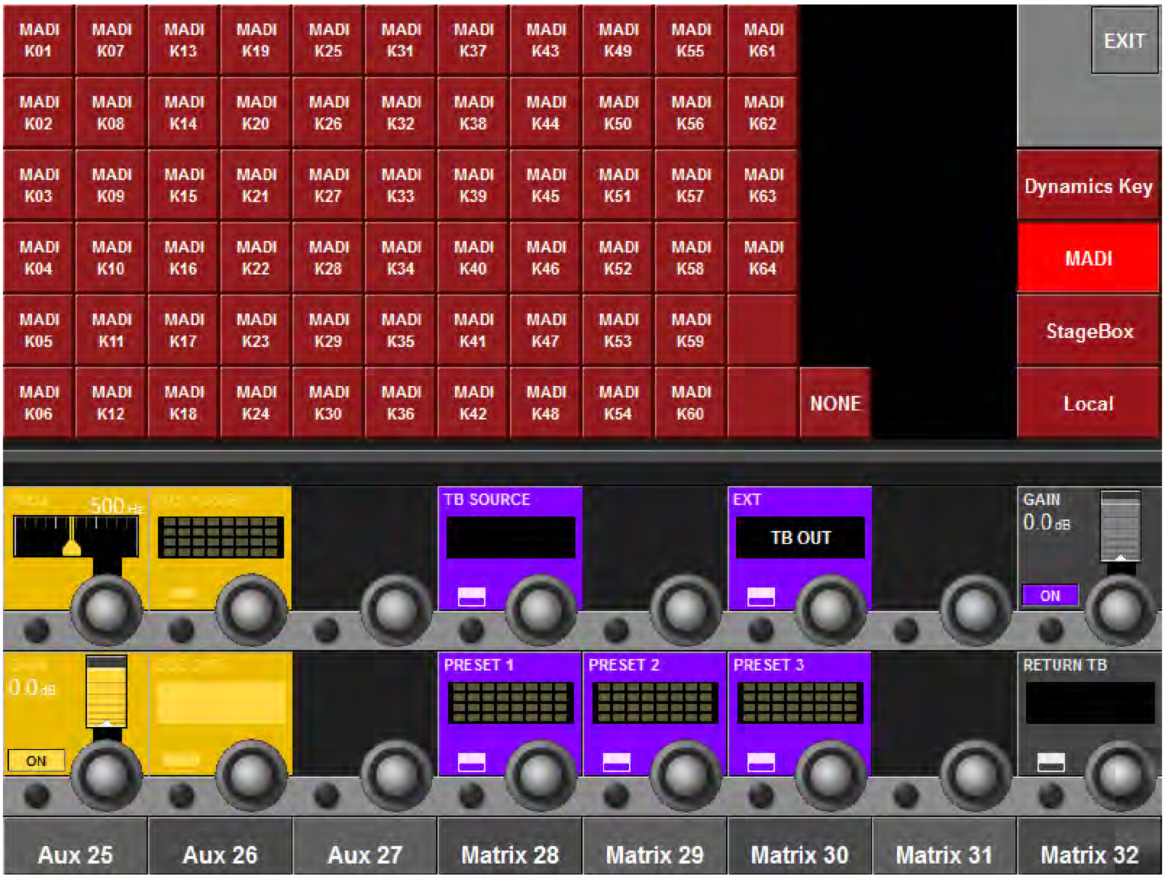


Figure 12-4: Oscillator Output Patch Page.

This page allows the user to patch the oscillator to outputs on the StageBox and the Local Rack, to MADI channels, or to the key inputs of the dynamics units on the input channels 1-64.

The user selects the appropriate screen by selecting one of the following touch screen buttons: <Dynamics Key>, <MADI>, <StageBox> or <Local>.

TB Send Section

This section allows the console operator to route the TB mic signal to various outputs of the Soundcraft Vi Series™.

TB Source Field

Displays the source name. If nothing is selected the console's TB mic XLR (parallelled) pair is automatically selected. Its {VST config button} opens the TB source patch configuration page (Figure 12-5).

HINT: Sources can be the internal TB Microphone or any Microphone Input from the Soundcraft Vi Series™. Select <NONE> to select the console's TB microphone XLR sockets.

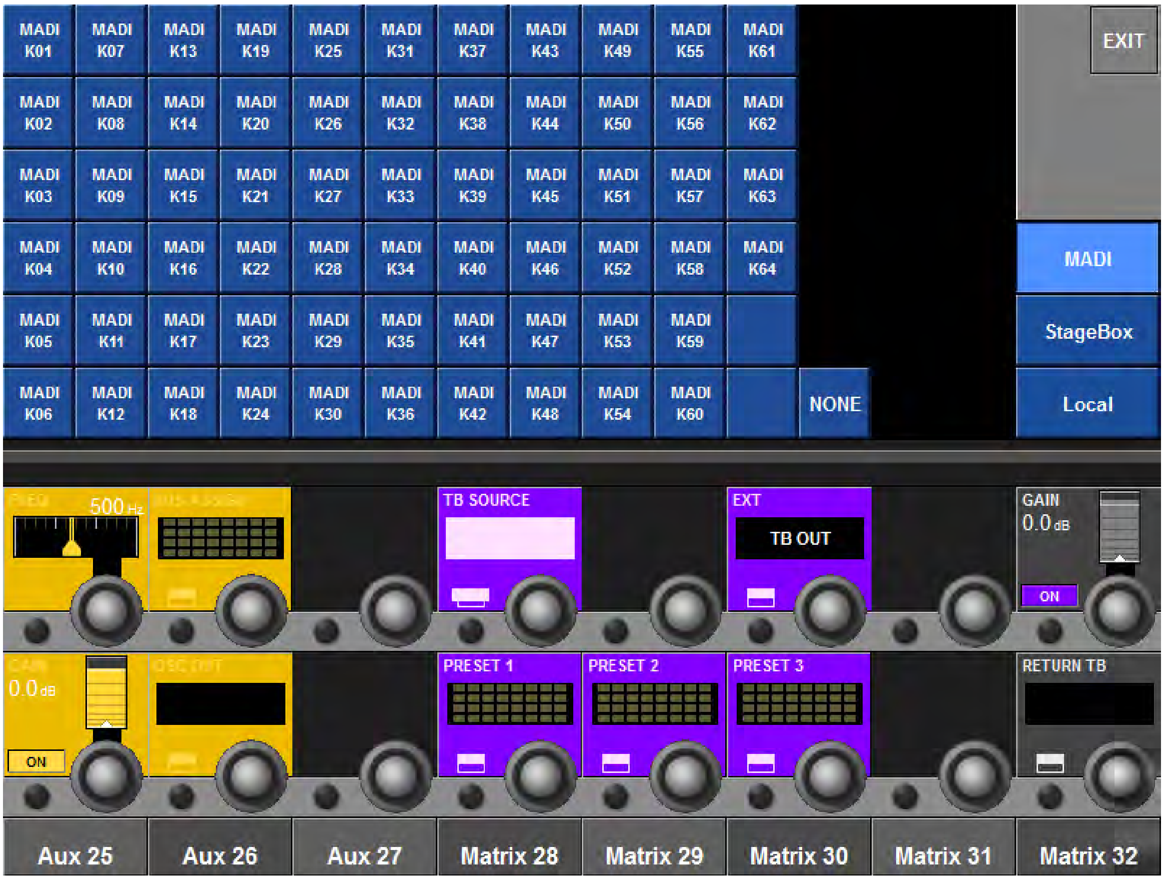


Figure 12-5: TB Source Configuration Page.

Ext

The {VST config button} opens the patch configuration page (see Figure 12-6). This allows the user to select which output will be used when the [EXT] key on the front panel is selected. The default is the TB OUT socket on the local rack.

Preset 1 – 3 Fields

Their {VST config buttons} opens the configuration page to setup the preset patches to the Busses (this page is similar to Figure 12-3).

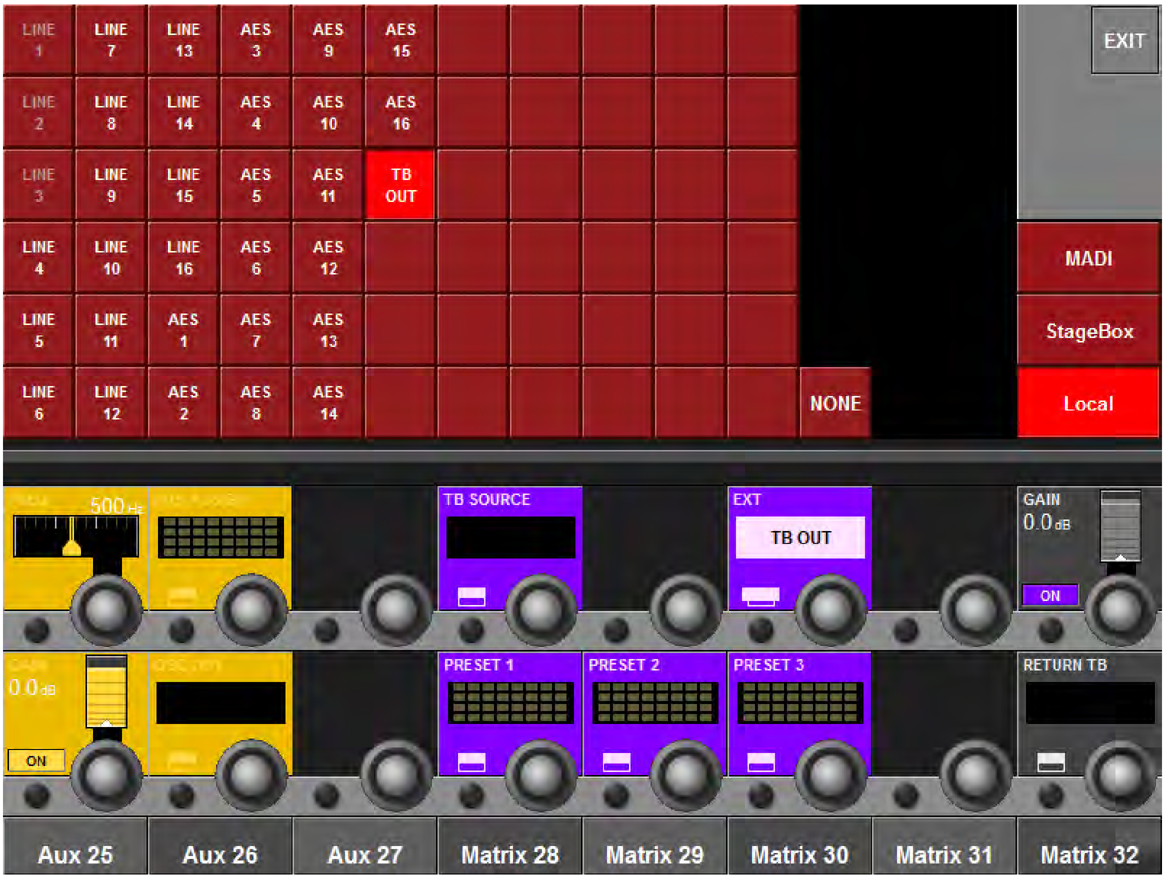


Figure 12-6: External TB Configuration.

TB Return Section

This section 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. This function must be enabled via the **monitor setup** page.

Gain

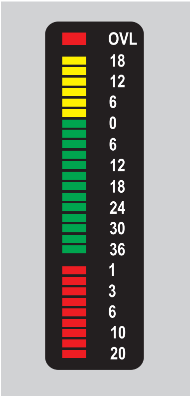
Encoder adjusts the TB return signal level. {ON} enables the TB return.

Return TB

Its {VST config button} opens the patch configuration page to choose the physical connector that will be used for the TB return signal (this page is similar to Figure 12-5).

Input Channel Meter

Each input channel has an input meter. There are two parts to the meter (see Figure 13-1): The top part is a 20--segment level meter, and the lower part is a 9-segment Gain Reduction Meter (GRM).

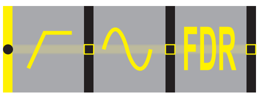


The level meter shows the input level at a selected point in the input channel. The point at which the input meter measures the signal within the input channel can be globally set via the Settings page, there are four options (see Figure 13-2). This page is accessed by pressing the [MENU] key, and then pressing the <Settings> tab.

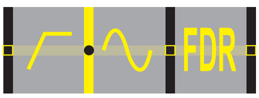
If two channels are vertically paired, the meter shows the higher of the two values.

The GRM shows the overall gain reduction of the limiter and compressor, if they are engaged.

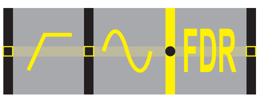
Figure 13-1: Meter Panel (Level Meter & Gain Reduction Meter)



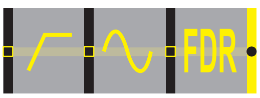
The Meter Point is after the analogue mic gain section, but before the digital trim and filters.



The default setting. The Meter Point is after the digital trim and filters, but before the Gate/EQ/Dynamics.



The Meter Point is after the Gate/EQ/Dynamics, but before the fader.

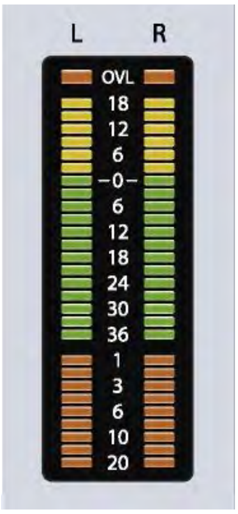


The Meter Point is after the fader.

Figure 13-2:Selecting the Input Meter Measuring Point

Bus Master Meters

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, but note that this feature is reserved for a future release).



The GRM shows the overall gain reduction of the limiter and compressor, if they are engaged. Note: the Dynamics from Stereo Bus Masters are always linked, therefore both GRMs show the same value.

Figure 13-3: Bus Master Meter Panel (Level Meter & Gain Reduction Meter)

Master Output 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 similar to Figure 13-3, and the C master has a mono version similar to Figure 13-1.

Monitor Meters

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

Scale

The Level meter scale goes from +18dB to -36 dB, it represents the actual output level in dBu from its analogue line output. Gain reduction is displayed in the Range 0 - 20 dB.

HINT: Soundcraft Vi Series™ contains full floating-point calculation, 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.

In the Input Channel meters, the overload (OVL) LED indicates an overloaded analogue input (mic preamp clipping), while in the Master meters the overload indicates a value that is higher than Full Scale (analogue output-stage clipping).

Meters On The Master Section Screen

The meter panel is always visible if [Meter Lock] is enabled, unless a SETUP or configuration is active. If [Meter Lock] is disabled and a Solo/Sel button from a master is active, the meter panel will be temporarily replaced by the master processing view.

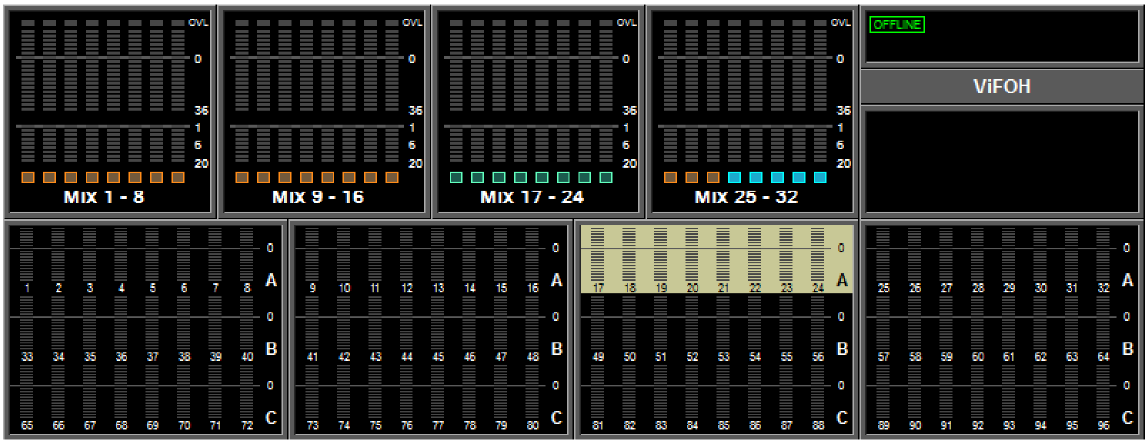
The upper region shows all 32 Bus levels, while the lower part shows all 64 input levels.
To enhance the overview, the input levels are split into two rows that correspond to the Channels in Fixed layer A (1-32) and Fixed layer B (33-64).

Stereo Busses are indicated by the two meters being joined at the lower end of the bar.
Paired Input channels are displayed with a white border, showing either horizontal or vertical pairing.

Figure 13-4: Meters On The Master Section Screen

Touch Selection

The input meter overview is also touch-sensitive and allows blocks of 8 input channels to be selected and assigned to the far right-hand bay of the console surface. A grey/white overlay indicates which, if any, block of 8 channels has been selected; the [A] or [B] key on the Input Fader Pages panel also flashes to indicate the temporary activation of the right-hand bay.



The selection can be cancelled in one of three ways:

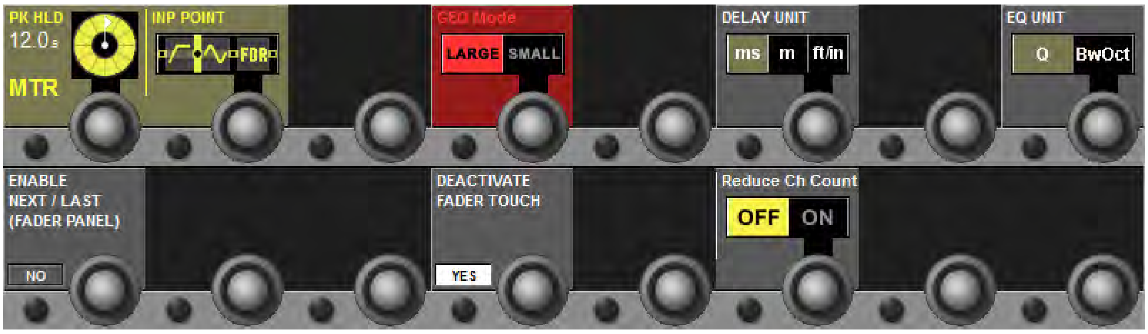
i) by touching another 8-channel block on the input meter screen,
ii) touching the same block again, or
iii) pressing one of the [A] or [B] buttons mentioned above.

This method is also used on the Vi4 to access the extra channels in a 96 channel system (see Chapter 4, Inputs).

Peak Hold

All Meters (LED Meters on the Surface and the Screen Meters) offer a PEAK HOLD function with auto release.

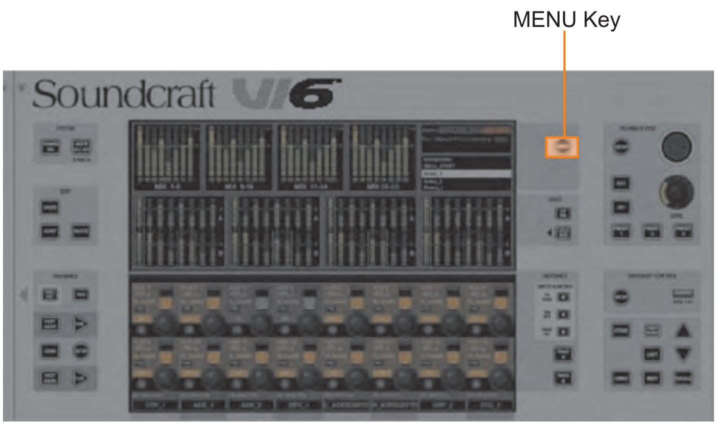
The Peak Hold time (same for all Meters) is adjustable via the SETTINGS Menu, and is adjustable from 0 to 12 seconds in 0.05 second steps. Setting to 0 effectively turns peak hold OFF.



Ballistics

The metering ballistics for all level meters is according to the PPM (peak program meter) DIN standard, with the difference that the attack time is audio sample based (20.8 uS @ 48 kHz) and has no integration time.

The GRMs have no ballistics, because they show the actual dynamics control value (with time constants according to those set by the user in the Dynamics VST page).



MAIN MENU

Figure 14-1: The [MENU] Key.

Pressing the [MENU] key opens the Main menu page(see Figure 14-2) in the master section's screen.

MAIN

Figure 14-2: Main Page.

The other menu pages can be accessed by touching the appropriate tab at the top of the screen. The software Release number and Build must be quoted when requesting technical support from Soundcraft personnel.

SECURITY Field

{LOCK} locks all switches, encoders and faders on the console, except fot the {LOCK} key.

BRIGHTNESS Control

The user has a choice of three brightness levels for the screens, illuminated keys and FaderGlow™.

SHOW

Figure 14-3: Show Page.

A complete explanation of Shows is given in chapter 15 of this manual.

GPIO

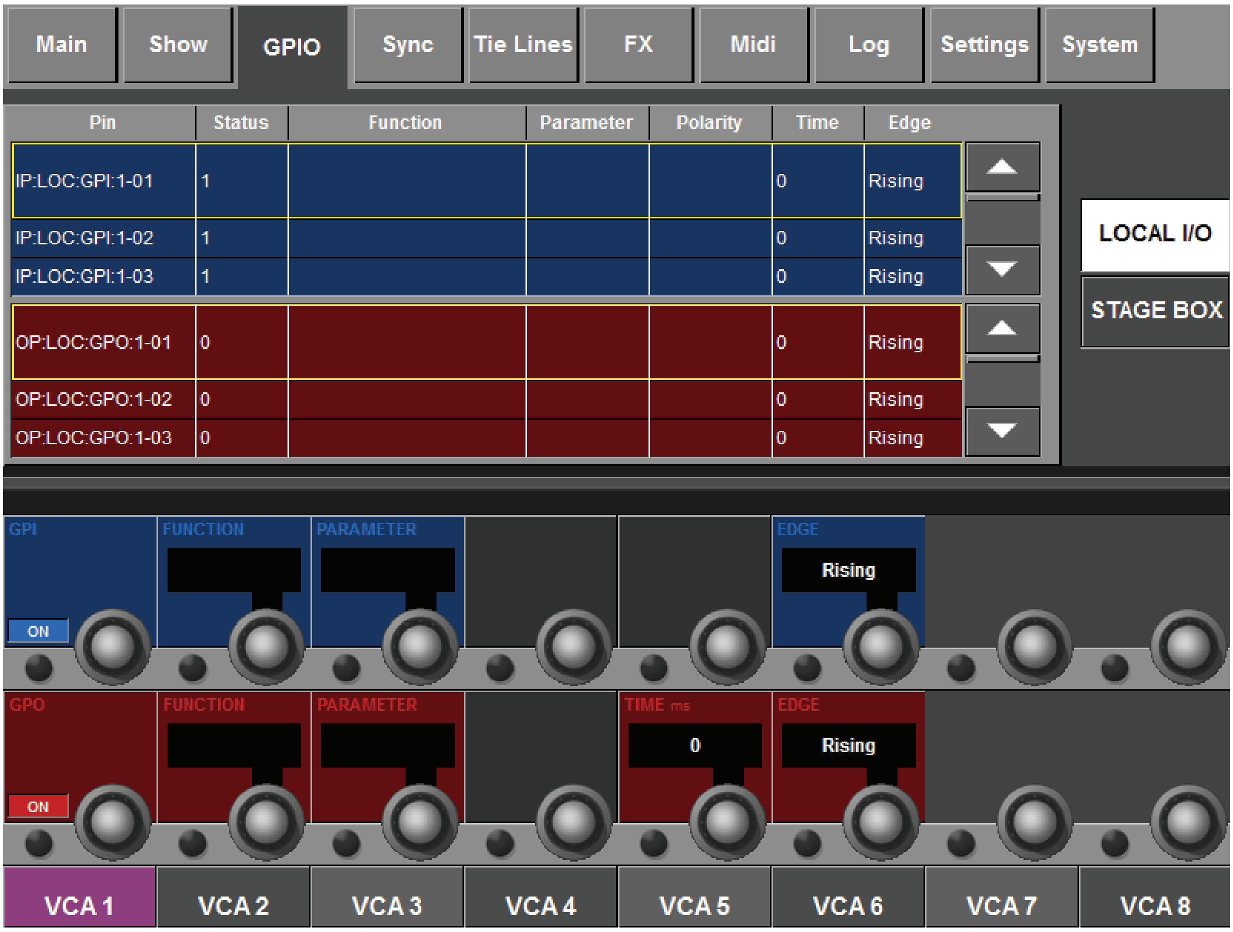


Figure 14-4: GPIO Page.

A complete explanation of GPIO usage is given in chapter 18 of this manual.

SYNC

Figure 14-5: Sync Page

Currently the console will work only on an internal clock of 48kHz.

If an external clock is connected to the clock card in the Local Rack (to the wordclock, AES or video sync inputs) the console will switch to external sync, and the EXT indicator, together with the WCLK, AESor VIDEO indicators (as appropriate) will illuminate.

In order for the console to lock to the external clock its frequency must be 48kHz +/-100ppm (+/- 0.01%).

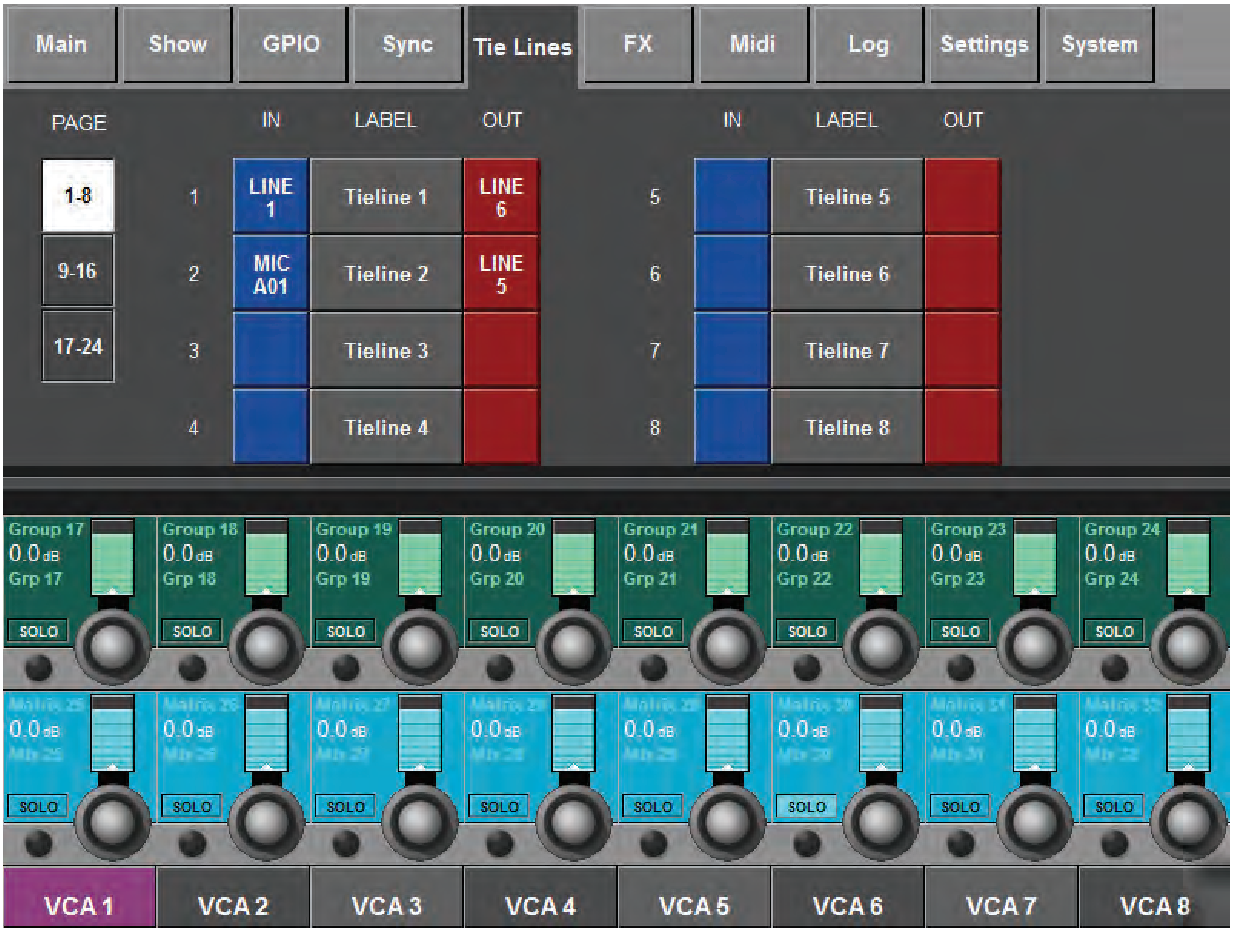


Figure 14-6: Tie Lines Page.

A complete explanation of Tie lines usage is given in chapter 11 of this manual.

FX

Figure 14-7: FX Page.

See Chapter 21.

MIDI

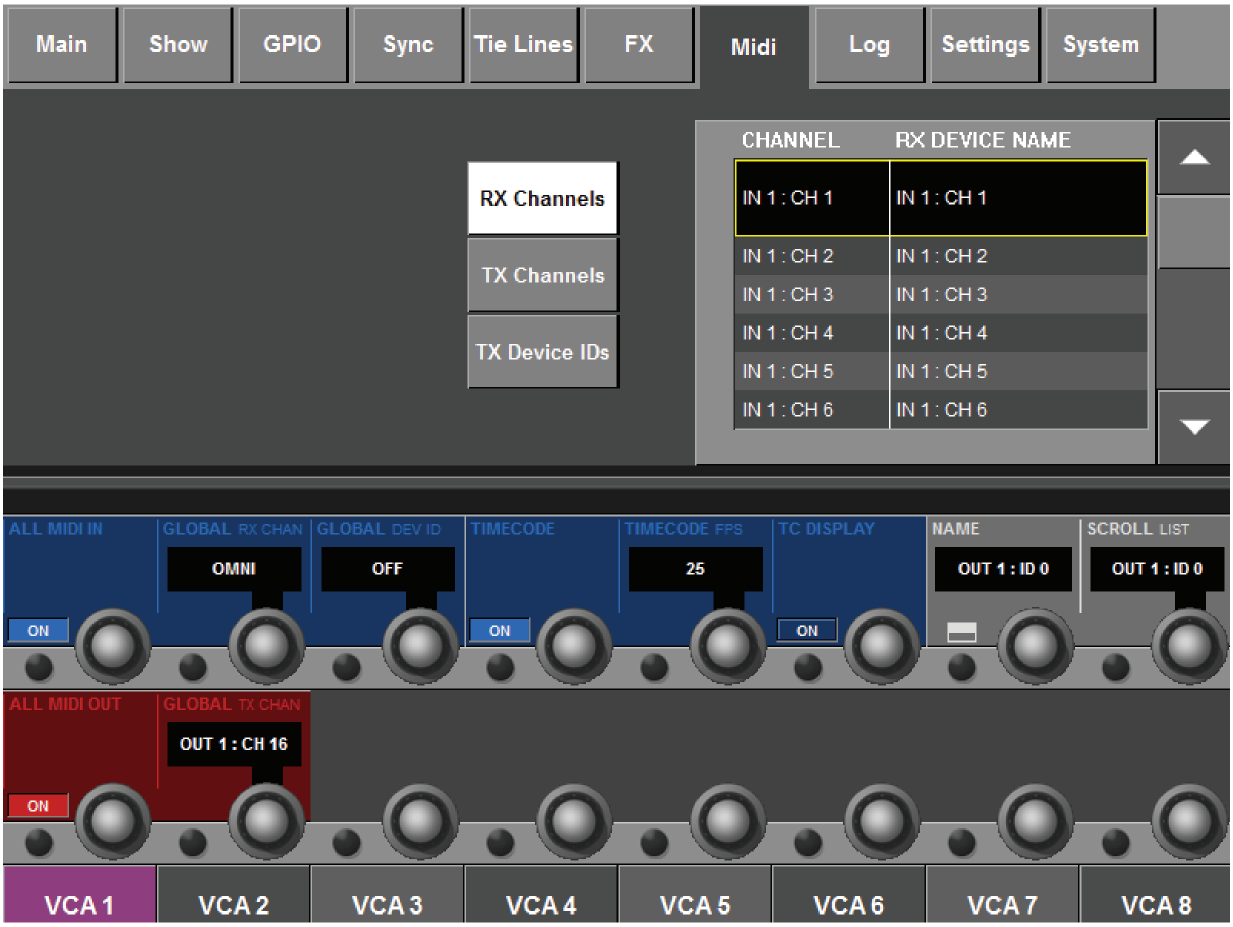


Figure 14-8: MIDI Page.

The MIDI page of the Main Menu contains the following elements :

- Device Lists for TX MIDI Channel, RX MIDI Channel and TX MIDI Device ID.
- Global MIDI Receive Channel, On/Off and Global Receive MIDI Device ID
- Global MIDI Transmit Channel and On/Off switch.
- MIDI Timecode RX global On/Off switch and Frame Rate control.

Further information is in Chapter 22.

LOG

Figure 14-9: Log Page.

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

The arrow butons allow the user to scroll up and down the list.
The <Expand> button displays 3 lines of text for the currently-selected message, the <Compact> button replaces <Expand>.

The <Auto Scroll> button causes the most recent message to be dispalyed as the currently-selected one.

The <Clear> button clears the log.

Hint: The most recent message is also displayed at the top right of the master section's main screen, in the Error Log Display Area (see Figure 14-11). 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.

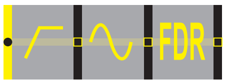
SETTINGS

Figure 14-10: Settings Page

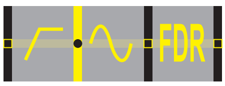
This page contains the general settings of the console.

POINT

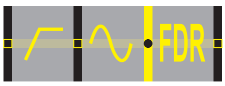
The point at which the input meters measure the signals within the input channels can be globally set via the {POINT} encoder. There are four options as shown below.



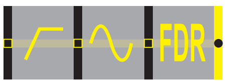
The Meter Point is after the analogue mic gain section, but before the digital trim and filters.



The default setting. The Meter Point is after the digital trim and filters, but before the Gate/EQ/Dynamics.



The Meter Point is after the Gate/EQ/Dynamics, but before the fader.



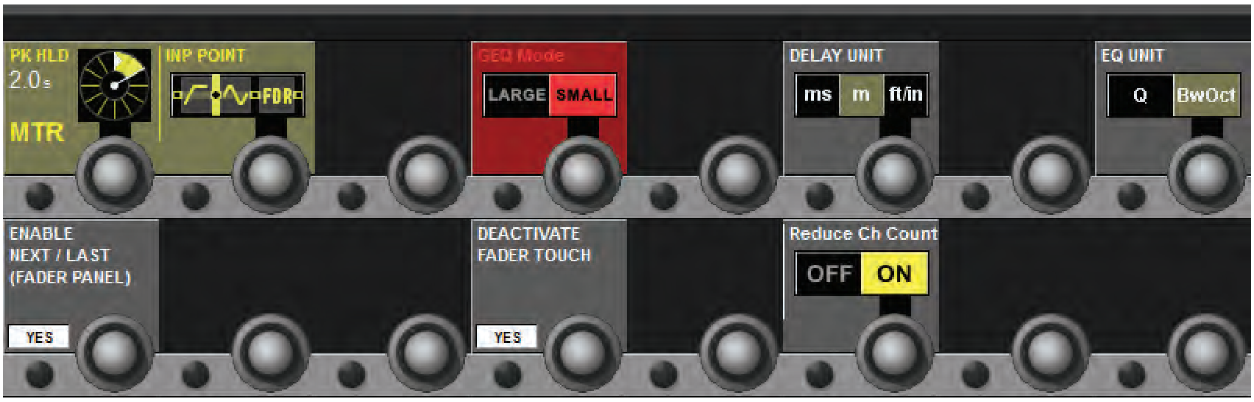
The Meter Point is after the fader.

ENABLE NEXT/LAST

The [NEXT] and [LAST] keys are duplicated near the front of the desk, above the [SOLO CLR] and [GANG] keys. For safety these duplicated keys are normally not enabled, and have to be switched on via the {NO/YES} key. Once the keys have been enabled this will be stored when the show is saved.

DEACTIVATE FADER TOUCH

If strong RF Fields are present (such as from a nearby MW radio transmitter), the operation of the fader touch sensors may be affected. The Fader Touch can be deactivated, to allow the faders to function without interference. The Fader Touch is set to deactivated as the factory default.



GEQ Mode

This mode determines how many faders are used on the desk to control the graphic EQ bands.

The ‘Large’ 30-fader mode allows fast access on multiple faders, at the expense of access to the input faders. (normal).

The ‘Small’ (8-fader) mode is used when the opertor requires access to input faders at all times. Frequency bands can be scrolled in banks of 4 or 8 bands, using the Output Fader page buttons.

DELAY UNIT

This control allows the current delay time value on inputs, outputs and monitors to be displayed as milliseconds, metres or feet & inches, allowing the operator to choose the most appropriate unit for setting up delay.

Distance conversion assumes fixed temperature of 20 degrees C/68 degrees F

The Delay unit setting is saved in the Show file.

EQ UNIT

This control allows the operation of the bandwidth controls in the EQ sections throughout the console to be selected as either Octaves or Q-factor. Until now this has been fixed as Q-factor.

The direction of the control is reversed between the two settings: In Q mode, clockwise narrows bandwidth, in Octaves mode, clockwise widens bandwidth.

The ‘Octaves’ setting provides a more intuitive control in a musical context.

The setting of the control is saved in the Show file.

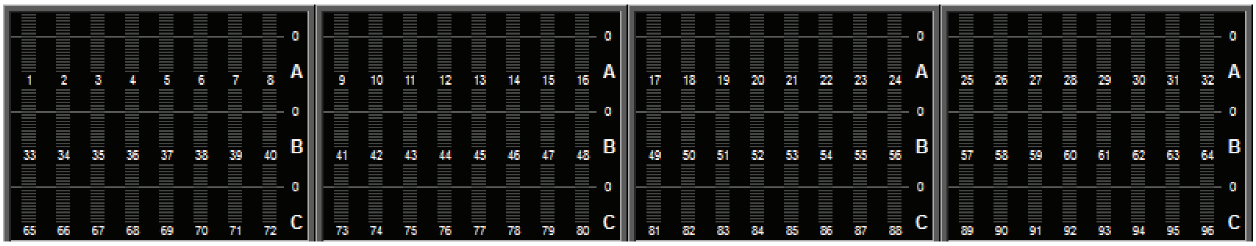
REDUCE Ch Count

When a third DSP card is fitted to Vi2, 4 or 6 consoles, the input channel count is increased from 64 to 96 channels.

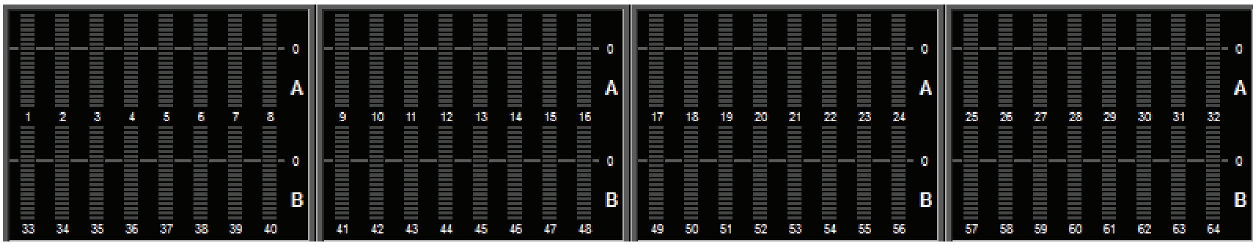
There may be occasions where even on a fully expanded console, more than 64 channels are not needed and in this case the channel count may be reduced back to 64 using this control in the Settings page.

The main benefit of this is to give easier handling of the touch-selection of blocks of inputs using the meter overview screen.

Input meter screen with REDUCE CH COUNT control OFF - 96 inputs, smaller meters



Input meter screen with REDUCE CH COUNT control ON - 64 inputs, larger meters



SETTING THE INTERNAL CLOCK

Press the Menu button on the console surface to open the Menu page. Choose the Settings tab at the top of the Menu page.

The current time/date setting of the console's internal clock is displayed at the top left of the screen. Touch the SETUP button on the touchscreen below the time and date display to open an editing page on the Vistonics encoders below. The encoders and buttons then allow various aspects of the time and date to be adjusted, as follows:

DAY, MONTH and YEAR encoders set the date.

DATE FORMAT encoder sets one of three date formats for the console: These are: DD/MM/YYYY, MM/DD/YYYY, YYYY/MM/DD. The console uses the selected format wherever it displays date information (eg: Show file creation dates, time display in Main Menu page).

HOURS, MINUTES, SECONDS encoders set the time.

TIME FORMAT encoder selects either 12-hour or 24-hour clock format.

AM/PM encoder selects AM or PM (12-hour time format only).

APPLY/CANCEL buttons applies the edited values to the console's clock, or cancels the edits and returns to the previously set date/time.

SYSTEM MONITORING

Overview

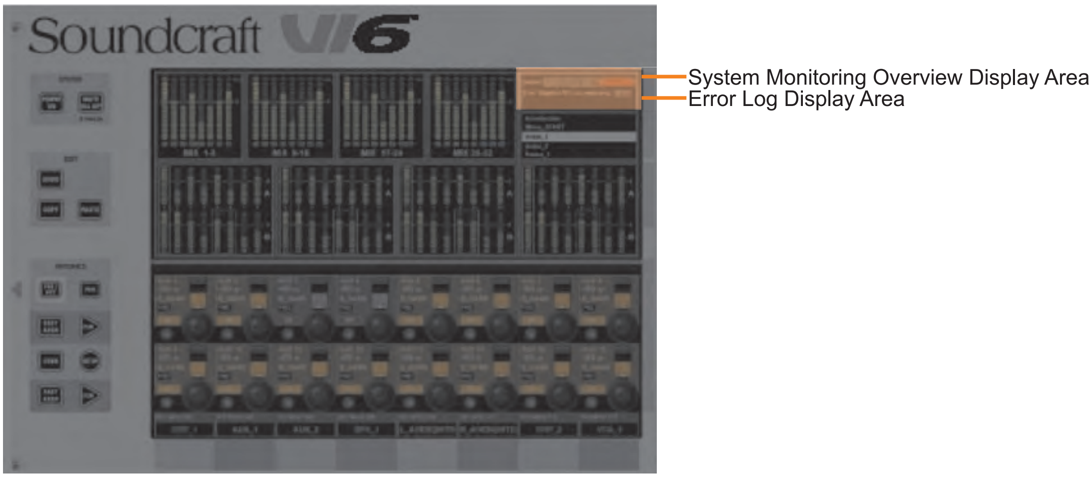


Figure 14-11: System Monitoring Overview and Error Log Display Areas

The System Monitoring Overview Display Area is located at the top right of the master screen, the Error Log Display Area is just below it (see Figure 14-11).

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.

HINT: in addition to accessing the System page by pressing [MENU] then <System>, the user can also touch the System Monitoring Overview Display Area.

When the System page has been opened, there are 4 sub-pages which are accessed via the touch-pads on the right hand side: DESK, LOCAL I/O, STAGE BOX and HiQnet™.

DESK

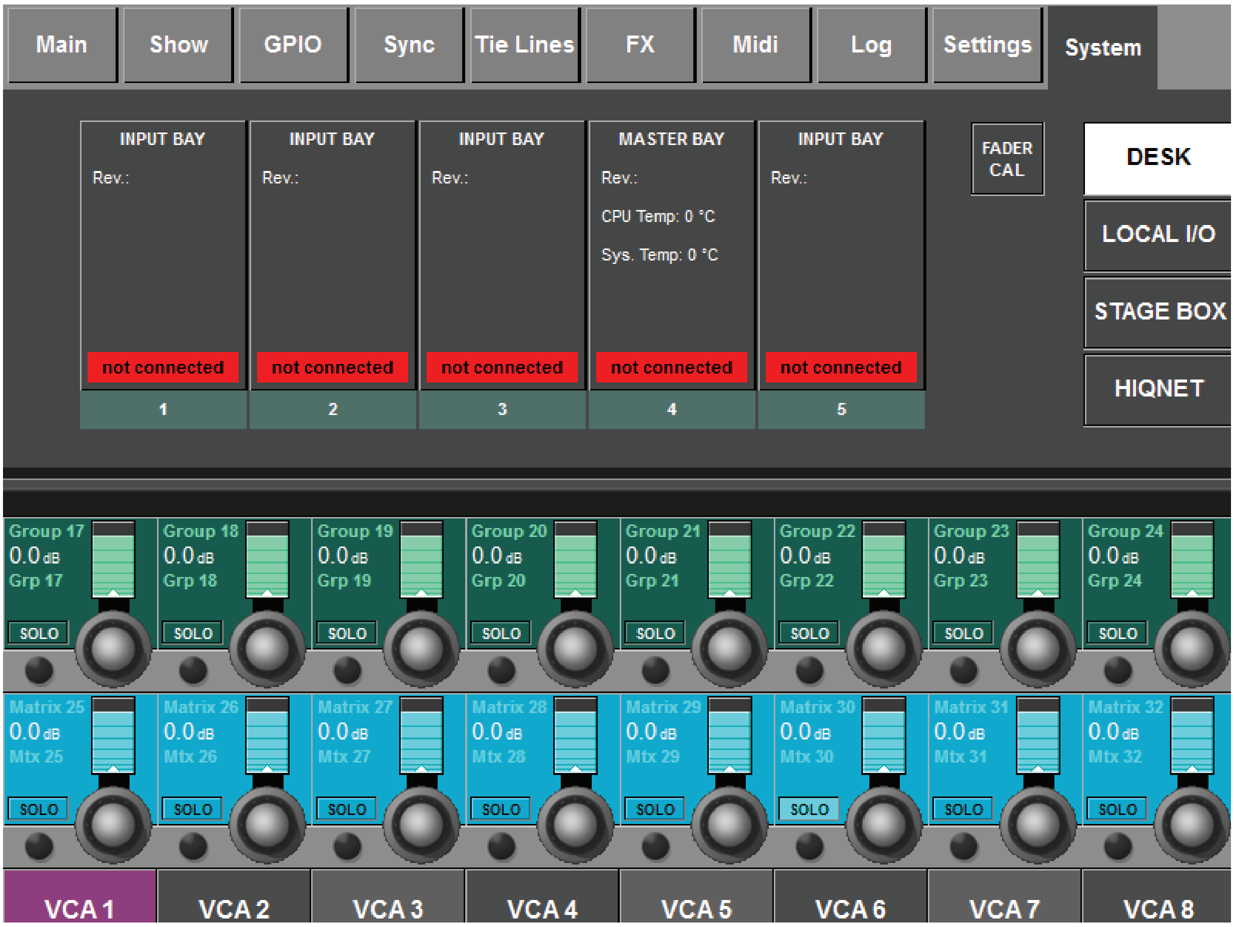


Figure 14-12: System - Desk Page

This page displays the current status of the desk's 5 bays.

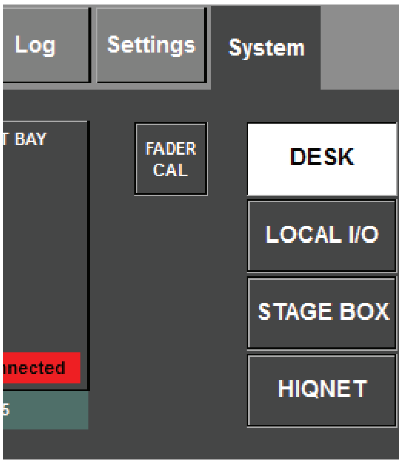
The numerical data displayed for each bay gives the revision number of the firmware currently installed in each bay. This information may be needed by Soundcraft service personnel if technical or service support is requested.

FADER RECALIBRATION

Under normal conditions, the faders of Vi2/4/6 will never need to be recalibrated. There is no requirement to do this on a routine basis. The only times this will be necessary will be either after the fader PCB has been replaced due to a fault, or liquid has been splashed into the faders, in which case it will be obvious that something is wrong with the behaviour of the faders.

The sign that faders need to be recalibrated will be either that paired channels or busses exhibit a 'creeping' effect where the faders move slowly by themselves up or down, or that the faders do not return to the same positions after changing layers (this can also be confirmed by watching the numerical values of the Bus Master faders which are displayed in the Control Bay whilst changing output fader pages.

Instructions for Recalibrating the faders



1. Navigate to MENU-System-Desk and press the FADER CAL button on the right hand side of the touchscreen. Answer YES to the warning dialogue box (you will lose control of audio for the duration of the process).

2. Follow the instructions that are now displayed in the Short Chanel Label displays – initially the faders will all move to the bottom of travel and the instruction will say "Set all faders to minus infinity". Although the faders may appear to already be at minus infinity, carefully move each fader knob so the cursor line is aligned with the minus infinity mark on the fader scale.

3. When all faders have been set in this way (do not forget the Master and Monitor faders in the Control Bay!), you will see a message in the Short Label displays saying "Press Any Key to Continue" - press any of the ON buttons on each Bay to start the lower end calibration process. The display will indicate 'Find Minimum Value' whilst this is happening,

4. When this has finished, the faders will all move to the top, and the message will be displayed "Set All Faders to +10". Repeat the procedure of carefully aligning the fader knob cursor with the +10 mark on the scale, and when you see the message 'Press Any Key to Continue", press an ON button in each bay to start the upper calibration. The displays will indicate 'Find Maximum value' whilst this is happening.

5. When all bays have finished, the faders will return to their previous positions and the calibration procedure is complete. The calibration will be permanently stored the next time the console power is shut down.

LOCAL IO

Figure 14-13: System - Local I/O Page

This page displays a graphical representation, and the current status, of the cards in the Local Rack. It 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 the card indicates that it equipped with inputs, a red label indicates that it is equiped with outputs. Some cards have both inputs and outputs, they are shown with both colours in triangles.

If the card configuration of the Local Rack is changed (e.g., when optional cards are fitted in place of existing cards), the card labels will be automatically updated with the new card types if the 'Reconfig' button on the Local Rack has been pressed after changing the cards.

If an optional card has been fitted and the word 'BLOCKED' is shown in place of 'OK', this means that the new card has exceeded the total number of allowed input and output channels within the system (192 inputs and 192 outputs). This can happen particularly if additional MADI cards are fitted. In this case it is necessary to limit the number of channels on the new cards by changing their DIP switch settings.

STAGE BOX

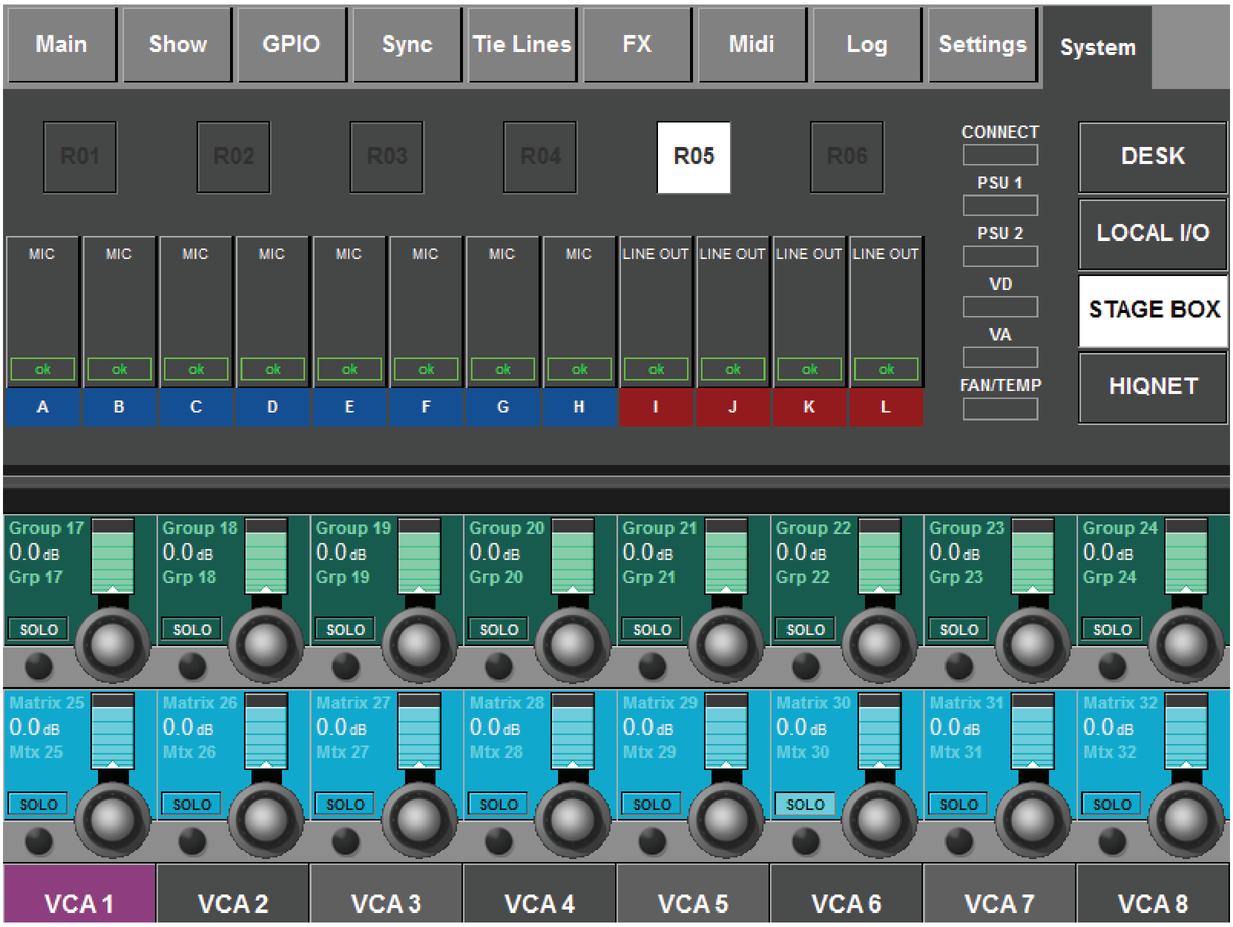


Figure 14-14: System - Stage Box Page

This page displays a graphical representation, and the current status, of the cards in the Stage Box. It 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 (e.g., 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.

The <R01> to <R06> buttons at the top of the screen are illuminated to show the connected Stage Box(es). These buttons relate to the valid double slots in the Local Rack where a MADI card could be connected. The default position when only one Stage Box is connected is R05.

HiQNET

Figure 14-15: Settings Page

HiQNet {ON}
Enables or disales the HiQNet ethernet port on the rear of the control surface.

HiQNet ADDRESS
Allows the HiQNet Address for the console to be edited. Every piece of equipment on a HiQNet network must have a unique HiQNet address.

IP CONFIG: DHCP or MAN
Set to MAN if you wish tio manually set the IP address for the console, or to DHCP if you wish the address to be assigned automatically by an external DHCP server (e.g. a network switch)

IP ADDRESS and SUBNET MASK
If the IP CONFIG mode has been set to MAN, the IP address and subnet mask controls allow these to be set for the console. In an ethernet network, every piece of equipment must have a unique IP (Internet Protocol) address. Devices taht need to communicate with each other must be on the same subnet as each other.

Note: when changing settings on the HiQNet page there will be a delay of up to several seconds before the change is actioned. This normal, and is due to the configuration time of the internal network interface hardware.

The Snapshot system allows the user to store records of the console's settings. When a Snapshot is stored it becomes part of a Cue: a Cue contains a Snapshot and optional MIDI and GPIO/HiQnet events. These Cues can then be recalled during a performance. Cues can be deleted, copied and moved within the running order of the show.

The Cues are stored on the console's flash drive, each set of Cues is stored as a Show. The Shows can be backed up onto, and downloaded from, a USB data storage device.

Note that some of the console's settings are not stored within Cues, but they are recorded as part of the Show. These settings therefore do not change within a Show. Other settings are not recorded at all. A list of what is recorded, and what is not, is provided at the end of this chapter.

SNAPSHOT FILTERING

Snapshot filtering means the selective recall of certain snapshot parameters. The complete set of parameters is always stored, so filtering only affects recall.

There are two types of snapshot filtering: Snapshot Scope and Global Filter.

Snapshot Scope is a way of selecting the parameters which are recalled from snapshots. The Snapshot Scope is stored with each snapshot. This allows special snapshots to be created to perform specific functions, only acting on a defined part of the console.

Global Filter is stored in the Show, and affects all snapshots. It can be edited and switched on or off using the ISO buttons on the channel strips.
Global Filter is useful as an 'emergency' tool to stop some parameters from being recalled, e.g. a microphone slipping making an EQ change necessary, or a fault of some kind meaning a spare channel has to be used which has all parameters set to defaults in all snapshots.

Any snapshot recall is subjected to both these filters in series (assuming Global Filter is switched on), so if a parameter is filtered in either or both of the filters then it will not be recalled.

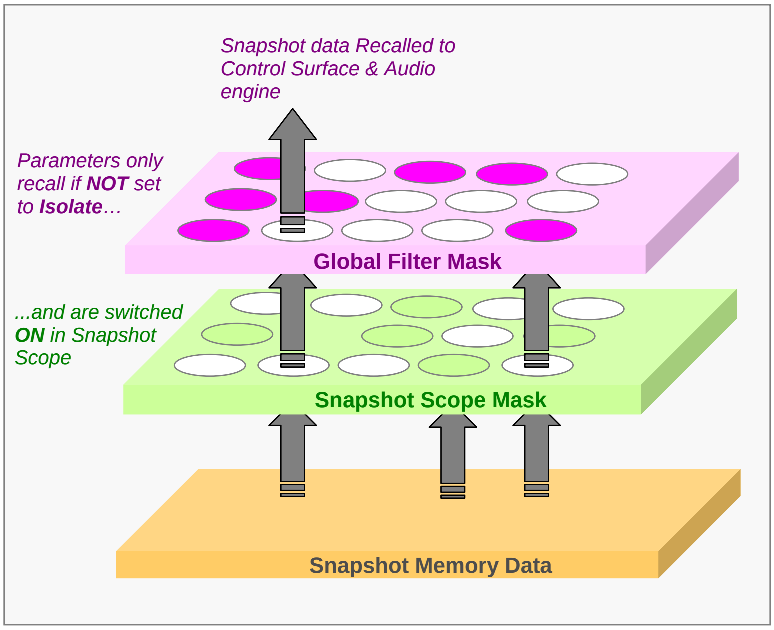


Figure 15-1: Showing The Effect Of Snapshot Scope And Global Filter On Recalled Data

FRONT PANEL DISPLAY AND CONTROLS

The Cues of the currently-loaded show are displayed in the Cue List displayed on the master screen (see Figure 15-2). The Show's title is also displayed.

Figure 15-2: Cue List.

The Snapshot/Cue keys (see Figure 15-3) are used as described below.

Figure 15-3: Front Panel Snapshot/Cue Controls.

SETUP

The [SETUP] key causes the Setup page to be displayed on the master screen, see Figure 15-4.

DATA Socket

This accepts a USB data storage device. There are three data sockets on the console: the other two are located on the rear panel.

STORE

Pressing [STORE] will record the console's current settings into a new cue.

UNDO

Pressing [UNDO] will undo the effect of pressing [RECALL], [NEXT] or [LAST]. It is useful if any of these 3 keys get pressed by mistake, and if the console's previous settings, which will get overwritten, are needed and have not yet been recorded into a cue.

PREV MODE
This mode locks the audio where it is, and allows the control surface to be recalled to any desired Cue, for checking purposes, or to make updates to future Cues, without changing any audio.

The surface re-synchronises with audio when Preview mode is switched off. See page 6 for more information.

LAST
The [LAST] key is used in conjunction with the information displayed in the Cue List (see Figure 15-3). Pressing [LAST] causes any settings pointed to by the previous Cue, in the Cue List, to be loaded into the desk, i.e. the desk is configured according to the settings held against the previous Cue. This Cue becomes the current one in the list.

NEXT
The [NEXT] key is used in conjunction with the information displayed in the Cue List (see Figure 15-3). Pressing [NEXT] causes any settings pointed to by the next Cue, in the Cue List, to be loaded into the desk, i.e. the desk is configured according to the settings held against the next Cue. This Cue becomes the current one in the list.

The NEXT and LAST keys are duplicated near the front of the desk, above the SOLO CLR and GANG keys. For safety, these duplicated keys are normally disabled, and have to be switched on via the MENU/SETTINGS page. Once the keys have been enabled this will be stored when the Show is saved.

The Arrow Keys and RECALL
The [up arrow] and [down arrow] keys are used in conjunction with the information displayed in the Cue List (see Figure 15-3). Pressing either of these keys will scroll up or down the list, without implementing the settings of any of the Cues which are being scrolled through.

When the required Cue is reached, the user presses [RECALL]: this causes any settings pointed to by the selected Cue to be loaded into the desk, i.e. the desk is configured according to the settings held against this Cue. This Cue becomes the current one in the list and the cue name is displayed in green text to indicate this.

Snapshot Crossfade

Snapshot Crossfade allows the recall of a desk snapshot to happen over a predefined time interval, rather than immediately. The interval can be set anywhere from 0.1 to 30 seconds, in 0.1s increments, using the Crossfade time control. This parameter applies to all parameters on all channels globally on the desk (it is not possible to set different Crossfade time on different channels).

Most ‘variable’ audio parameters of the desk that are included in snapshots will be included in the Crossfade, the exceptions are listed below.

- EQ and Hi/lo cut frequencies
- All Lexicon FX parameters

All switched parameters, plus the exceptions listed above will have their values changed at one of three points in the Crossfade: at the start, in the middle, or at the end. This is set globally for all the parameters via the ‘Switches’ control.

To set the Crossfade time for a specific Cue, press the Setup button in the Snapshot Control area of the control surface to open the Cue List page, then select the required Cue using the scroll bars or up/dwn arrow keys, and touch the name area of the currently selected Cue in the centre of the Cue List (the area becomes highlighted white, as shown in the picture below):

The time selected on the XFADE control below the Cue List will be the time taken for the desk to change from its current state, to the state of the snapshot in the selected Cue. In other words, the XFADE time can be thought of as an 'In' time for the Cue.

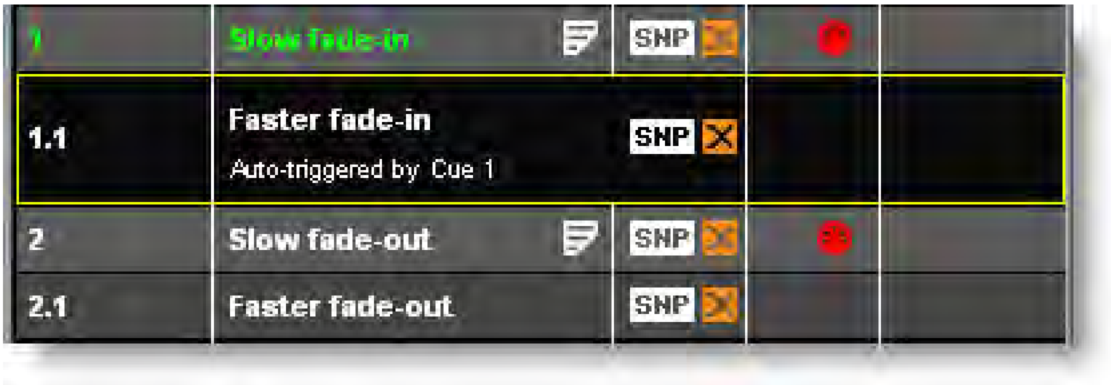
Each Cue can have its own 'In' time set using the XFADE control.

The Crossfade time can be disabled without affecting the time by using the On/Off switch. An icon  is shown in the Cue List next to the desk snapshot icon if a Crossfade time has been enabled for that Cue.

Using Crossfade and Cue Chaining to create 'pseudo-dynamic Cues'

Using the Cue Chain facility that was part of the Version 3.0 software, in conjunction with the Crossfade function allows an approximation to 'dynamic' cue fader automation to be achieved. Using the Snapshot Scope facility to control what is recalled on each Cue can also be used to achieve different Crossfade times on different channels, if that is required.

To do this, try to break up the overall fader move required into several sections, and make Cues corresponding to the start and end points each section. Then chain the Cues together using the 'Go To Cue' parameter in the Cue List. (To find these parameters, open the Cue List page and touch the left-hand side of the currently selected Cue bar in the list):



Old Show files and Crossfade parameters

When loading an old pre-V4.0 Show file, be aware that these do not contain any Crossfade information in their Cue List, and so whatever Crossfade settings that might be present on the desk, for example from the previously loaded Show, will remain on the desk and appear to get copied into the Cue list of your old show.

To avoid this, always load a Default show onto the console before loading a pre-V4.0 show that has not yet been saved on a V4.0 desk. Loading the Default show will clear out any Crossfade parameters and result in the cue list of the old show loading correctly. Once the old Show has been saved on a desk running V4.0, the problem will not happen any more, with this particular show.

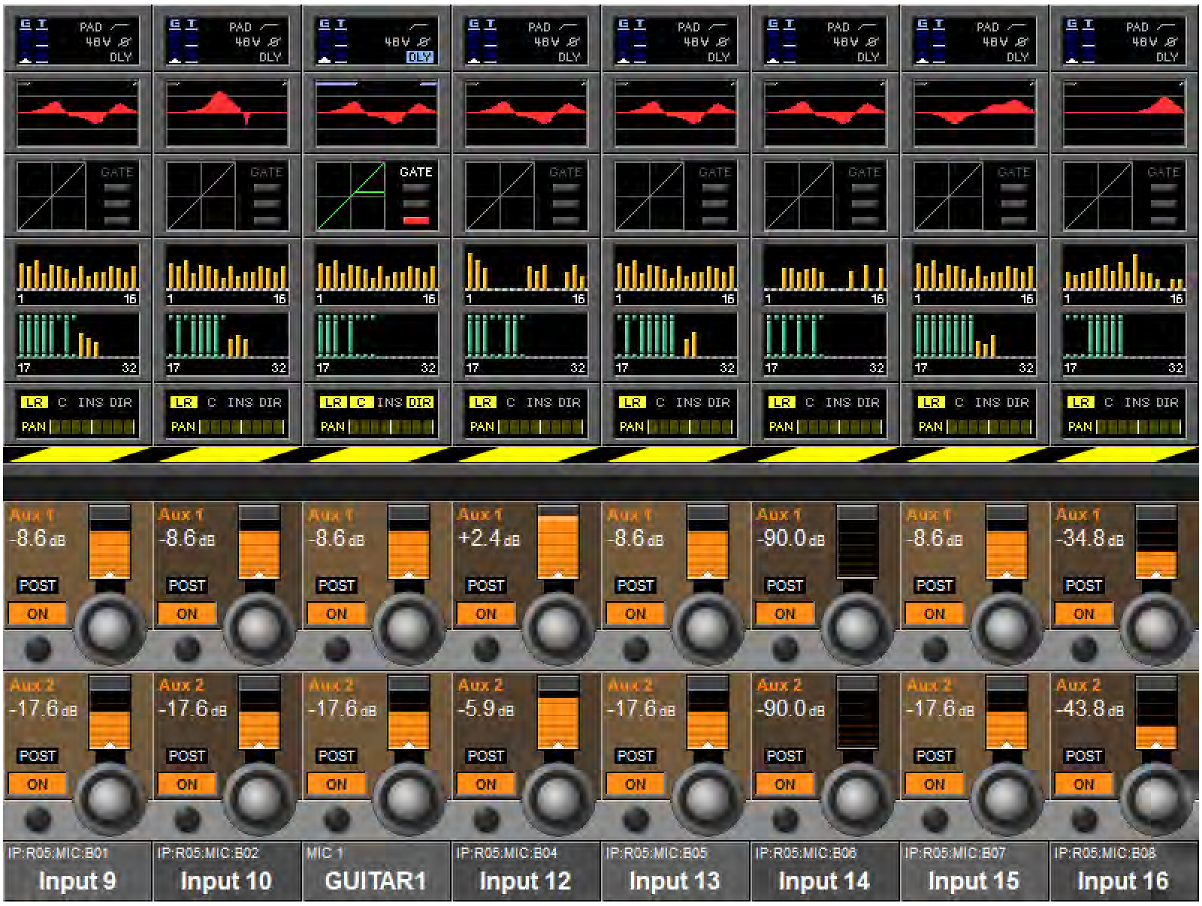
Snapshot Preview Mode

Snapshot Preview mode allows snapshots to be recalled to the console surface without affecting the audio running in the DSP core, and so provides as useful way to check what is about to be recalled in a Cue, during a show.

When the desk is in Preview mode, the Control Surface is effectively taken off-line from the DSP core, so that existing Cues can be recalled or edited, or new ones created, and there will be no effect on the audio, which will continue running with the settings that were active at the moment Preview mode was switched on.

When Preview mode is switched OFF again, the surface will automatically jump back to match the state that it was in at the moment Preview mode was switched ON – meaning that it will once again be in sync with the audio.

When the desk is in Preview mode, no control of audio is possible, so the Preview button itself flashes, and a yellow/black striped strip is displayed across the bottom of all Input bay touch screens as a warning.
If you have made changes to the desk parameters whilst in Preview mode, and you want to keep these,



you must either update an existing snapshot, or create a new one, otherwise the changes will be lost when you exit Preview mode.

You can also use Preview mode to ‘lock’ the audio before you change to a new Show file – the audio settings will remain as per the old show, and when you switch off the Preview mode the new Show settings will be applied to the audio.

Cue List Display

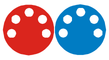
A Cue is a combination of a desk snapshot, various types of events and some text notes. The list allows the snapshots to be combined with transmitted events and arranged into a running order.

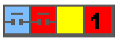
The Cue List is stored in the current Show. The list comprises columns for Cue Number (or Timecode), Cue Name, Desk Snapshot status (DESK), MIDI event status (MIDI), and GPIO, HiQnet and Blackout event status (GPIO/misc).

The oversize entry with a yellow border, in the centre of the list, represents the currently selected Cue (this cue is not necessarily the currently recalled cue). Within the selected Cue is also an additional space that displays text notes that can be entered to give information about what the Cue does.

The columns of the list are able to display various icons that represent the various types of event that are possible to trigger, or be triggered by, the cues:

Snapshot icon: this is displayed in all Cues to represent the presence of the desk snapshot in the Cue, but is greyed out if the desk snapshot is disabled locally within the Cue. It also appears greyed-out if the Desk column has been globally disabled.


MIDI IN & OUT Icons: these icons indicate the presence of a valid MIDI event in the cue. The blue icon represents a MIDI In event, where a MIDI message can be used to recall the cue. The red icon represents one or more MIDI out event(s). These icons will be greyed out if the MIDI column is globally disabled, or if the MIDI IN and/or MIDI out have been switched off in the MENU\MIDI page.


GPI/GPO/Blackout/HiQnet event icons.
Displayed in the relevant column, these icons indicate whether an incoming or an outgoing event has been set for the Cue. If no event has been set the icon does not appear. Icons appear greyed-out if the column is disabled. In general, blue icons indicate incoming events, and red or yellow icons indicate outgoing events.

Global Enable/Disable control using Cue List Column heading buttons
These column headings can be touched to disable an event column (all events of that type are disabled for the whole list) or the whole Cue List can be completely disabled by touching the Cue #/Cue Name column heading. Pressing the column headings has a toggle disable/enable action. As with all other parameters on the Cue List page, the states of these buttons is stored with the current Show.

<Cue List> column heading button: Enables/disables the recall of entire Cue List. If set to Off, the Cue List is locked and Cues cannot be recalled. The scrolling of the list, and editing of events is still allowed: only the [NEXT], [LAST] and [RECALL] keys on the surface are disabled.

<Desk> column heading button: Enables/disables the recall of desk snapshots for all Cues.
When set to off, no desk snapshots will be recalled, even if desk snapshots are switched On within individual Cues. If a new Cue is created (e.g. by pressing STORE) whilst the Desk column is disabled, a cue will be created which does not include any desk snapshots.

<MIDI> column heading button: Enables/disables both incoming and outgoing MIDI events for all Cues.
When set to off, no MIDI messages will be sent or received, even if the MIDI events are switched On within individual Cues.

<GPIO/Misc> column heading button: Enables/disables both incoming GPI and outgoing GPO events for all Cues. When set to off, no GPIO messages will be sent or received, even if the GPI or GPO events are switched On within individual Cues. For HiQnet, no HiQnet messages will be sent, even if the HiQnet events are switched On within individual Cues. Any blackout events set within cues will also be disabled.

SETUP

Pressing [SETUP] opens the following page. This page can also be opened by touching the Cue List in the top right corner of the Master screen, see page 15-2.

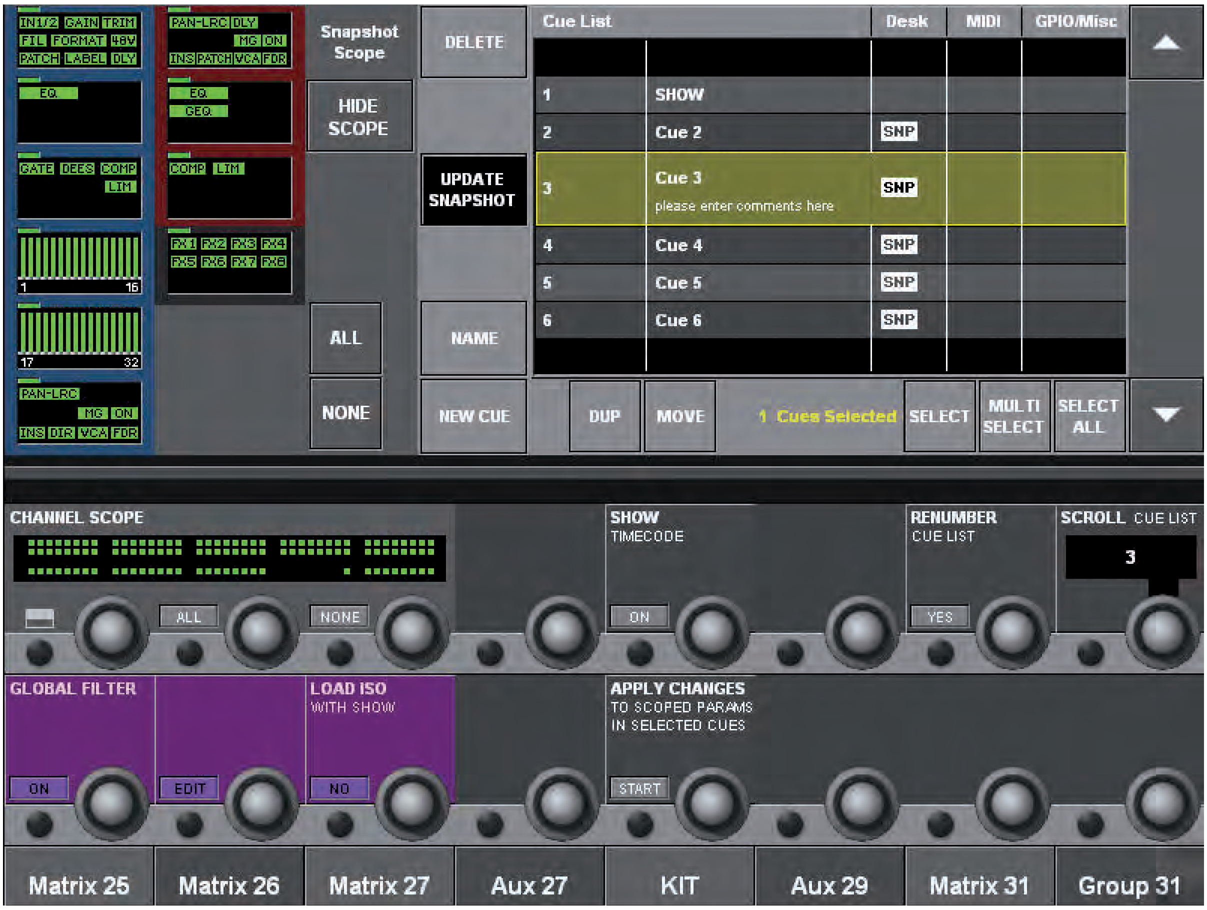


Figure 15-4: Snapshot & Cues Setup Page.

HIDE SCOPE/SHOW SCOPE Buttons

If the left-hand side of Figure 15-4 is not shown on your console it is because the Snapshot Scope is hidden in order to simplify the screen for new users. If you wish to see these controls, press the <SHOW SCOPE> button. The setting of the show/hide scope is stored in the Show.

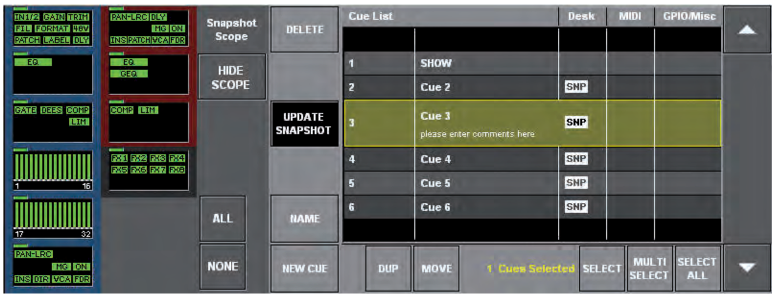


Figure 15-5: Cue List Touch-Screen Layout Map

The Cue List main page is divided into three main sections, please refer to Fig 15-5: The Cue List display itself (tinted grey for clarity in Fig 15-5); a set of touch buttons (tinted pink) associated with the Cue List and performing various editing and other operations on the Cue List; and the Snapshot Scope Graphical User Interface -GUI (tinted green), providing a way of quickly enabling various parameter groups to be recalled by desk snapshots, for each Cue.

Edit & Control Buttons

These touch buttons allow a) creation of Cues, scrolling of the list, and b) various editing operations on the list such as Delete, Move, Duplicate, rename etc. To allow these operations to be carried out on more than one Cue at a time, a set of Multi Select buttons c) are also provided.

a) Action buttons

<SCROLL UP/DOWN> buttons: Moves the Cue List up and down through the central selection cursor. These buttons are duplicated by the action of the UP/DWN arrow buttons on the surface. The Cue List cursor will always indicate the Cue that will be recalled if the Recall button on the surface is pressed. The main Cue List page also contains a green highlight indication which shows which was the last recalled Cue.

<NEW CUE> button: If Cursor is at the end of the Cue List, pressing New Cue creates a new Cue with default name Cue xxx, where xxx = number of existing entries in the list +1. If the cursor is in another position in the list, a new Cue is inserted in the next position in the list (see Cue Numbering). In both cases, a desk snapshot is also generated and associated automatically with this cue, and the Desk Snapshot enable/disable state is set to ON (enabled).
The events status of all other event types is set to disable (OFF) and no events are assigned.
The settings of the Snapshot Scope are also stored with the currently displayed settings, along side the audio and surface parameters.

<UPDATE SNAPSHOT> button: Updates only the Desk Snapshot associated with the Cue, by overwriting the snapshot with the current state of the surface. A dialogue box appears to confirm this action. Note that Snapshot Scope settings do not need the Update button to be pressed in order to save them - changes to Scope are stored immediately.

b) Edit function buttons

<NAME> button: opens the QWERTY keyboard to allow the name of the currently selected Cue to be edited. The Cue is given a default name on creation in the format ‘Cue X’. (The number is automatically incremented with each subsequent creation operation).

<DELETE> button: deletes the currently selected Cue(s). A dialogue box appears to confirm this action.

<DUP> button: creates a copy (or copies) of the currently selected Cue(s). The copies contain all aspects of the Cue, ie: Desk Snapshot and Events.
The names of the copies have a (D) added to the beginning of the Cue Name, in order to distinguish them from the originals, and are placed after the original.
If multiple non-adjacent Cues are selected and the DUP operation performed, the duplicates appear after their own original.

<MOVE> button: (Latching) Simulates a ‘click & drag move function. When latched ON, a pre-selected Cue or adjacent group of Cues can be moved within the list by using the scroll buttons or encoder.
Note that only a continuous range of Cues can be moved – the Move operation will be inhibited if a selection of non-adjacent Cues is active when Move is pressed.

The <MOVE> button is renamed <DROP> after it is switched on, and pressing <DROP> will 'drop' the Cue or group of Cues at the point immediately after the last visible Cue above the selection bar. The Cue Numbers of these moved Cues will be recalculated according to the 'Inserted Cue' numbering rules (see Cue Numbering).

c) Selection mode buttons

<SELECT> button.
The SELECT button allows single or any number of adjacent or non-adjacent Cues to be selected for Delete, Duplicate or Move operations. It is equivalent to CTL+Click on a PC running Windows.

Touching SELECT will change the background colour of the central Current Cue selection bar from black to pale yellow. The Cue can be deselected by pressing Select again.
If the list is scrolled to another Cue after one has been selected, the yellow background will be retained on that previously selected Cue. A new Cue can now be chosen in the central bar and Select pressed again to add this one.

To deselect selected Cues, each one must either be brought into the Current Cue selection bar one at a time and the SELECT button pressed to deselect, or a 'select all' followed by 'select none' operation can be carried out –see Select All below.

<MULTI SELECT> button: dabbing the <MULTI SELECT> button selects the currently highlighted Cue with a latching mode, as with Select described above, but in this case the scroll control or arrow keys can be used to scroll through the list, and a continuous range of cues will then be selected.

When the required number of Cues has been selected, the Multiselect button is switched OFF, and the selection range stays in operation – the range can be seen by all visible selected Cues having a pale yellow colour.

Another range of Cues, not necessarily adjacent to the first, can be selected by repeating the above procedure in a different part of the Cue list. The 'Selected items' field (see later) keeps track of the number of selected Cues and is useful when some of the selections are outside the visible window.

Deselecting a range of Cues in Multiselect mode
If any of the Cues in an existing selected range is positioned in the Current Cue selection bar, then Multiselect is switched ON, the whole existing range selection is cancelled, ready to select a new range.

In order to go back and deselect individual Cues, the <SELECT> button must be used, as described above.

<SELECT ALL> button: Selects ALL the Cues in the List. When all Cues are selected, the button changes to a <SELECT NONE> button, which when pressed, deselects all Cues.
Pressing this button twice can therefore be used as a shortcut to clearing any existing selections in the Cue List.

Cues Selected field: A number is displayed next to the Select button, indicating how many of the Cues are currently selected.

CUE NUMBERING

CUE NUMBERING

New Cues that are created at the end of the list (depends on cursor position when Store or New Cue is pressed) are always given whole numbers.

Inserting cues (by moving an existing one or creating a new Cue with the cursor in the list) always generates a new number with one or two decimal places, at approx the mid-point of the existing numbers:

1.0
  Insert -> 1.5
2.0

1.0
  Insert -> 1.25
1.5

MOVING CUES

Moving Cues causes the moved Cues to be automatically renumbered.
The following example shows what happens to the numbering when Cues 2 & 3 are moved one step up the Cue List.
Select the range, press <MOVE>, scroll to required position, press <DROP> (cues will be renumbered),

1	cue 1	Move Cue 2 & 3 one step up:	**1**	cue 1
2	cue 2		**4**	cue 4
3	cue 3		**4.3**	cue 2
4	cue 4		**4.6**	cue 3
5	cue 5		**5**	cue 5

If some of the numbers end up being duplicated after this operation, this can be solved by selecting a wider range than the original block and pressing the Renumber Cue List button.

DUPLICATED CUES

A duplicated cue counts as a new cue as far as numbering is concerned.

RENUMBERING CUES

If cues have been moved around or have been inserted in the Cue List, the Cue Numbers will be a mixture of whole and decimal numbers. The RENUMBER CUE LIST {YES} key will renumber the cues. Pressing the button initiates a renumbering of the Cue List (Cue # column), the cues are renumbered as consecutive integers.
NOTE: There is a confirmation box which displays, 'Are you sure you want to renumber the Cue List?' and displays <YES> and <NO> touch buttons. The operation cannot be undone.

MANUALLY RENUMBERING

The Cue number can always be manually edited at any time by pressing the NAME button to open the QWERTY keyboard for renaming the Cue. The top left field in the keyboard allows the Cue number to be selected and a new number typed in. The Cue will be moved to the appropriate place in the list, according to the number given.

Snapshot Scope GUI

Please refer to Fig 15-6. Note the <HIDE SCOPE> button, if the screen on the Vi isn't showing the scope information, press the <SHOW SCOPE> button.

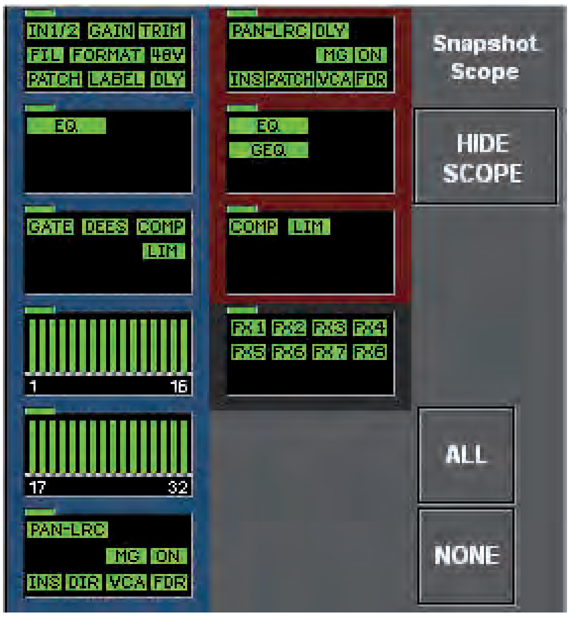


Figure 15-6: Snapshot Scope GUI.

The Snapshot Scope GUI allows the Snapshot Scope filter to be edited globally – by function block or parameter group, and by channel. Shortcut buttons in the GUI allow <ALL> or <NONE> of the parameter groups to be quickly selected or deselected in the Scope.

The elements displayed are divided into three categories: Input Channel (blue block border), Output Channel (red block border) and FX (grey block border). Within these categories, the parameters are grouped by function blocks.

Each Cue has its own Snapshot Scope, and the state of the Snapshot Scope for each Cue can be seen by viewing the Snapshot Scope GUI whilst scrolling through or recalling the Cues. **Note that the Cues do not have to be recalled to do this.**

The Scope of each Cue can be easily edited by simply touching the Scope fields, to select a whole block, or pressing & holding, to zoom a block onto the VST encoders below, where the encoder touch or VST buttons are used to select parameter groups within the block.

Illuminated Green mini-icons in the GUI show which parameter groups are enabled for recall in the corresponding snapshot. Note that these mini icons do not necessarily correspond exactly to the ones in the actual channel strips, but instead represent Parameter Groups, e.g. in the Dynamics block there is a Gate parameters group which represents the individual channel parameters of threshold, attack, hold, release and range: these individual parameters cannot be enabled for recall individually, they have to be selected as a group of parameters.

In the disabled state, the mini-icons are displayed in a low-intensity colour corresponding to their function (eg: blue for inputs, green for dynamics, etc), but this changes to bright green in all cases when the parameter group is enabled.

At the top left of each function block there is a green indicator LED. It indicates if all, some or none of the parameters in the block are selected.

Storing changes to Scope Settings

Storing changes to the Snapshot Scope is NOT done in the same way as for other desk parameters: ie: it is not necessary to press the <UPDATE SNAPSHOT> button to save the changes made to Scope into the currently selected Cue. Instead, the changes made to the Scope are immediately saved to the Cue as they are made.

When a new Cue is created, by touching the <NEW CUE> button, the Snapshot scope settings used for the new Cue will be the same as are currently displayed in the Scope GUI. They can then be edited after the Cue is created.

Changes to Scope across multiple Cues

If more than one Cue is selected, using the Multiselect function, then the Scope parameters behave as if they were in a gang – changing a Scope parameter in one Cue will cause the same parameter to change in the other selected Cue(s).

As with normal ganging on the console, Scope parameters that were already set to the desired state of the parameter being changed will not change unless the parameter is changed back to its original state again, in which case they will follow.

Selecting a complete function block:
A short press on the function block will select all the parameter groups within the block – all the mini icons will change to bright green. As many function blocks as required can be selected at the same time.

Zooming into a function block:
Press & hold the function block to open the zoom page on the VST encoders below, and allow individual elements within the block to be toggled. A white border to the block indicates its zoomed selection state. To exit, press & hold again.
When enabled, both the VST field and the mini-icon in the function block change to bright green.

Channel-wise selection of Scoped parameters
Immediately below the Scope GUI, there is a section of the VST encoder that allows a channel selection (input and output) to be defined for the parameter groups. See Fig. 15-7.

Figure 15-7: Channel Scope GUI
The channel selection section works in conjunction with the parameter group selection in the scope GUI so that a parameter will only be enabled in the snapshot recall on a given channel if both the parameter and the channel are enabled.
The channel selection is therefore stored with each Cue, as with the parameter group status.
When a new Cue is created, the default state of the channel selection is ALL ON.

This channel selection GUI is designed to allow an overview display of which channels are selected, using a 'dot-matrix' type display. As the Cue list is scrolled, it is possible to watch the dot-matrix display and see which channels are selected on which cues, in the same way as the parameter groups can be watched in the upper part of the Scope GUI.

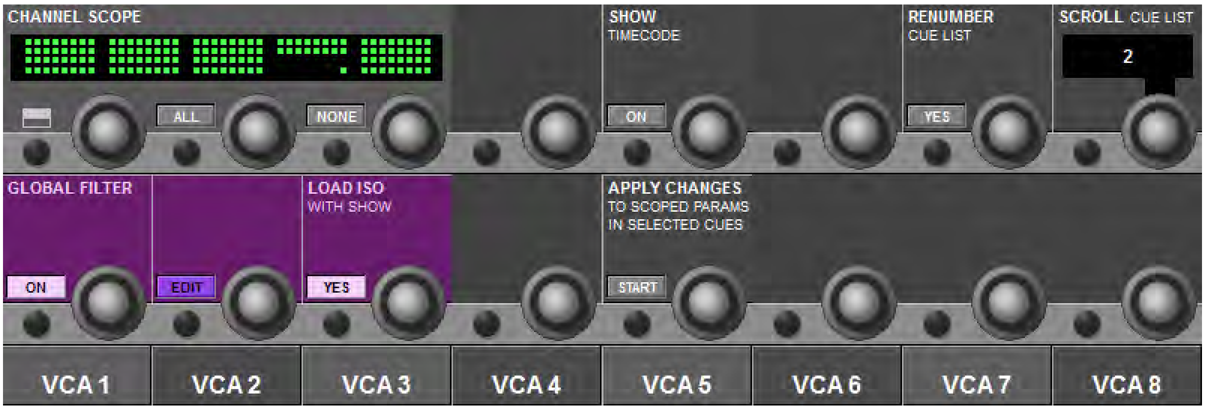
The  button is used to open a sub-page (see Fig. 15-8) that allows channel selection to be made on the touch screen for each of the Input Fader pages A-C and the All Busses page. These are selected by Bay number. Bay Numbers corresponding to Input Fader Bays give access to all input channels plus all busses, via the A,B,C and ALL BUSSES rows of 8 latching enable buttons.
The <ALL> and <NONE> buttons in the VST section allow quick setting and clearing of all channels, busses and VCA masters, without opening the sub-page. Note: depending on how many DSP cards are fitted to the desk, not all Fader Pages may be used.

Figure 15-8: Channel Scope Sub-Page

The bay number corresponding to the output master bay gives a different-looking sub-page that only displays enable buttons for the 16 VCA groups (2 rows of 8 buttons) and the LCR button, plus ALL and NONE buttons. (Note that this scope relates only to the VCA **Master** parameters (Fader & On), not to the channel assignments which are controlled within the input channel scope).
The **SOLO/SEL** buttons of channels and busses act as an additional way of selecting channels whenever the Channel Scope sub-page is open (Solo operation is always suspended when the sub-page is open).

SCROLL CUE LIST

The SCROLL CUE LIST encoder is always shown at the bottom right of the Cue List, and provides a faster alternative to scrolling the list using the scroll bars on the right-hand side of the touchscreen.



SHOW TIMECODES

When activated, the SHOW TIMECODE <ON> key replaces the cue number in the left-hand column of the cue list, with the trigger timecode value, if one has been set (see Page 15 - 13).

SORT BY TIMECODE

If a number of cues have a trigger timecode value set, pressing the {SORT BY TIMECODE} key allows the cues to be automatically re-arranged into timecode order. Cues that have no timecode trigger values are placed together in numerical order at the end of the list.

APPLY CHANGES TO SCOPED PARAMS IN SELECTED CUES

The software includes the capability to copy control settings that exist on the surface, into one or any number of other Cues in the Cue list. It is possible to define which controls from the current surface state will be stored, and which Cues you want to update with these control settings.
The changes made to the Cues will be 'absolute', ie the original setting of that parameter will be replaced by the new setting.

Start the process by ensuring the parameters you want to Apply are active on the surface. They do not have to be stored already in an existing Cue.
Press the START button in the Appy Changes field that is located on the bottom row of Vistonics controls below the Cue List.

Figure 15-9: Start.

This brings up the Scope selection panel on the left of the Cue list - this is now used to choose which parameters on the surface you want to copy to other Cues. You will notice the Scope panel appears with all parameters deselected, but all channels selected, this should speed up the selection process.
In the example (Fig 15-10), the EQ on channel 24 only has been selected.

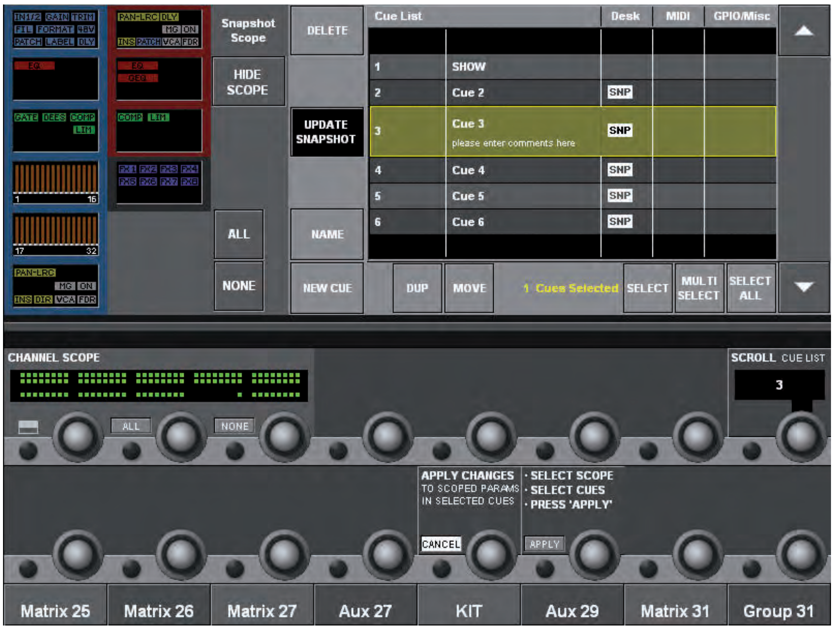


Figure 15-10: Scope and Channel Selection.

After the Start button has been pressed you will also see an additional field appear to the right of the Start button, giving the basic instructions for Apply Changes, and showing an APPLY button.
Once you have selected the parameters in the Scope selection panel, select which Cues you want to update by using the Select, Multiselect or Select All buttons below the Cue List.
Then when you are satisfied with your selection, press the APPLY button. You still have another chance to make changes or cancel the process at this point, because a dialogue box appears asking you to confirm that you want to update the snapshots in the selected Cues.
Touch YES to finish the operation or NO to go back to the selection stage. When you press YES, the desk goes through an automated process where the Cues are recalled and automatically updated. You will see a progress dialogue as this is happening.
Note that you will see controls moving on the surface as this is happening, but ***no audio will be changed*** during the process.

CUE LIST PAGE - Cue Number Field Touched

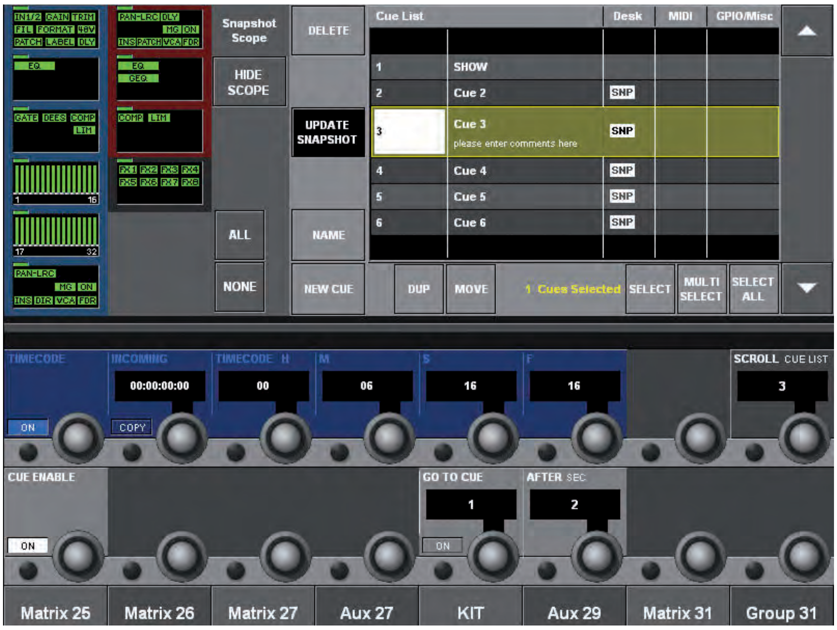


Figure 15-11: Cue Number Field Touched.

Touching the Cue Number area of the highlighted Cue switches the Vistonics encoders below to a new mode that allows specific parameters relating to this Cue to be edited.

CUE ENABLE {ON} key: Enables/disables the recall of this Cue in the List. When set to OFF, the Cue is jumped over in the list, if the [NEXT] and [LAST] keys are used to sequentially recall Cues. The Cue can still be selected by the scroll or up/dwn arrow keys on the surface (to allow editing of properties) but cannot be recalled by pressing the [RECALL] key (ie: no desk snapshot recalled, no events generated).
When set to off, the whole Cue appears greyed out in the Cue List.

SEQUENCER
The sequencer functionality allows auto-triggering of another Cue at a preset time delay after this Cue has been recalled. In this way several Cues can be 'chained' together.

GO TO CUE {ON} key: Enables/disables the sequencer function.

GO TO CUE encoder: Selects the Cue number of the Cue that will be triggered after the set time interval. (The numbers in the field match the current Cue List numbers).

AFTER SEC: encoder: Sets the time delay after which the Cue specified in the GO TO field is recalled. Range 0- 30s, in 0.5s steps.

TIMECODE encoders: Allows a MIDI timecode value to be set (Hrs:Mins:Secs:Frames). The Cue will be recalled automatically when the set timecode value is received at the MIDI In, if the **ON** button is enabled. There is also a global Timecode ON switch in the Menu \MIDI page which must be enabled in order for timecode triggering to occur. Timecode frame rate is detected automatically, and the value of the received timecode is displayed in the INCOMING field..
The {COPY} key transfers the value in the INCOMING field to the encoders.
The **{STEAL}** key becomes active if the whole Timecode value matches one that is already assigned to another Cue. Pressing {STEAL} will reassign this value to the current Cue. The default value of all fields will be '-' (= no value). The values will be shown greyed out if they match the values of another cue.

CUE LIST PAGE - Cue NAME Field Touched

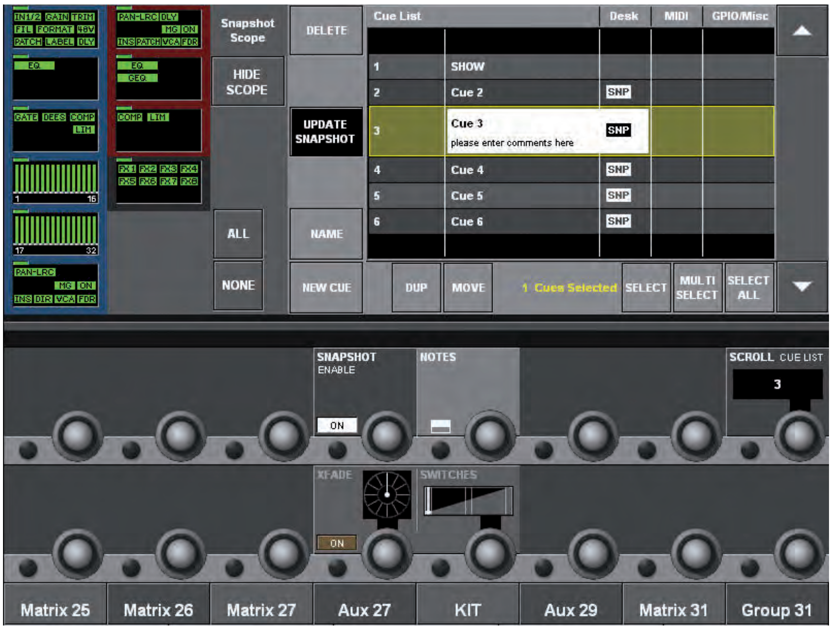


Figure 15-12: Cue Name Field Touched.

Touching the Cue Name area of the highlighted Cue switches the Vistonics encoders below to a new mode that allows specific parameters relating to this Cue to be edited.

SNAPSHOT ENABLE - {ON} key: Enables/disables recall of the desk snapshot in this Cue.
When not set to ON, the SNAP icon in the Cue List is greyed out.

NOTES sub-page key: Opens/closes the QWERTY keyboard and allows text notes to be typed that will be displayed in the highlighted Cue field in the Cue List page, and also (in shortened form) in the message area above the Cue List display in the Main Control Bay VST screen.

CUE LIST PAGE - MIDI Field Touched

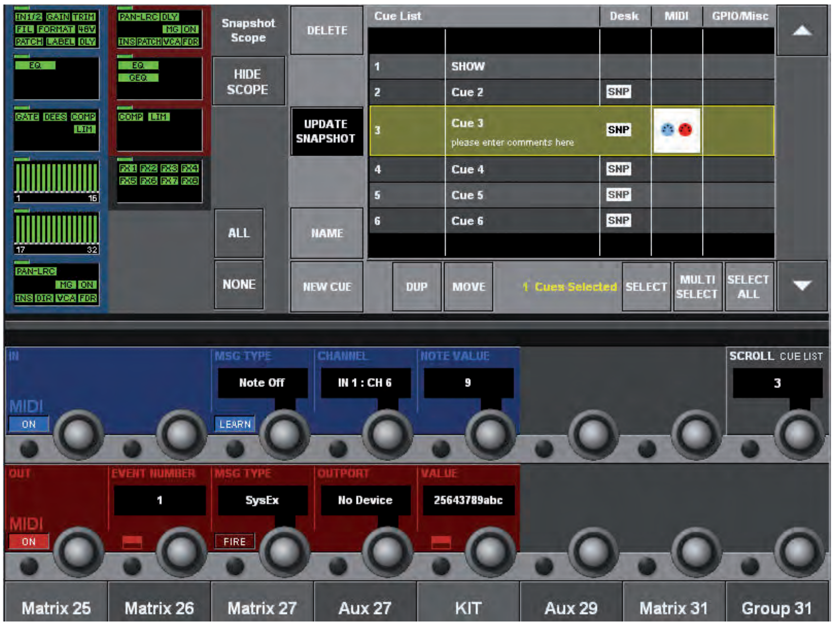


Figure 15-13: MIDI Field Touched.

This page displays setup controls for a unique MIDI IN message that can trigger the current Cue to be recalled, and up to 20 MIDI OUT events that can be sent when the current Cue is recalled.

MIDI IN Setup

{ON} key: enables/disables the selected MIDI parameters from triggering recall of this Cue when they are received by at the MIDI input. When off, the currently selected Cue cannot be triggered by an incoming event. The ALL MIDI In {ON} key in the Main Menu\MIDI page must also be enabled for messages to be received.

MSG TYPE encoder: Defines the type of MIDI Message being received for this Cue. See page 22-5 for Message types, how displayed, and whether used for RX, TX or both.
VALUE 1 encoder: Sets the Value 1 for the selected message type (field name may change dynamically to reflect actual parameter type according to selected message type).
VALUE 2 encoder: Sets the Value 2 for the selected message type (field name as above).
Value 2 is not applicable to all types of messages, but in this case an empty field will be displayed.
CHANNEL encoder: Allows the MIDI 'listening' channel for this Cue recall to be set. Value range is 'No Device', then 1-16, then 'Global' but the displayed value for 1-16 is taken from the Device Name field of the MIDI RX Device List (see Page 22-1, Main Menu\MIDI page). The Device list allows MIDI Channels to be mapped to a text name for easier identification of the devices being selected.

{REC} key: When active, the MIDI input of the desk 'listens' to incoming messages on all channels, and when it receives the first one that matches a supported trigger event type, it automatically populates the Channel, Msg Type and Value 1& 2 fields to match the received message. Any previous parameters are overwritten with no warning. The {REC} key switches off automatically when a valid message has been received, or the MIDI page is closed. (note: SysEx, MMC or MSC messages are not supported by the REC function).

{STEAL} key: Only appears if exactly the same combination of message type, channel, and values 1,2 (all these parameters must match) has been set up as a trigger on another Cue.
Pressing the {STEAL} key when it is visible immediately reassigns the displayed parameters to the current Cue.

Default settings stored in current Show are:

On key = OFF
Channel = 'No device'
REC key = OFF
Message Type= blank
Value 1&2 = blank

MIDI OUT Setup

The MIDI Out setup differs from MIDI In, in that instead of only 1 event for the Cue, there is an 'Events List' per Cue of **up to 20** events that can be transmitted.
Each of the events can be transmitted on any of the 16 MIDI channels on the two MIDI OUT ports (32 channels in total).

{ON} key: enables/disables the transmission of the currently-selected MIDI event when the currently selected Cue is recalled (separate value for each of the 20 events).
The ALL MIDI Out {ON} key in the Main Menu\MIDI page must also be enabled for messages to be transmitted.

EVENT NUMBER encoder: Scrolls through the 20 available MIDI OUT events on each Cue, and allows the Event parameters to be viewed or edited on the other VST encoders.

EVENT NUMBER sub-page key: Opens the Event List sub-page (see Figure 15.16). Although the 20 Events can be set up parameter-wise using the VST encoders, and scrolling the Event Number, the Events List makes it easier to see a glance all 20 events, and view their parameters in a table format.

MSG TYPE encoder: Defines the type of MIDI Message being sent for the selected Event Number.
See Chapter 22 for Message types, how displayed, and whether used for RX, TX or both.
VALUE 1 encoder: Sets the Value 1 for the selected message type.
(Field name may change to reflect actual parameter type according to selected message type).

There are two special cases of the MSG Type parameter – MMC Locate and SysEx – which cannot be accommodated by the Value 1 & 2 fields.

These two cases also result in the Channel parameter being replaced by a DEVICE ID, selected from the global TX Device ID list in the Menu\MIDI page).

When the **MMC Locate** message type is selected, the Value 1 & 2 fields are replaced by 4 fields allowing the Timecode value to be set – see Fig 15-14.

Figure 15-14: MMC Locate Message Type.

When the **SysEx** message type is selected, the Value 1 field changes to include a sub-page key which opens the QWERTY keyboard and allows the string to be entered in Hexadecimal format. When the string is longer than the number of characters displayable in the VST field, it is abbreviated with '...' – see Fig 15-15.

Figure 15-15: SysEx Message Type.

CHANNEL encoder: Allows the MIDI 'transmit' channel for the currently selected Event Number to be set. Value range is 'No Device', then OUT1:1-16 and OUT2:1-16, then 'Global', but the displayed value for OUT1:1-16 and OUT2:1-16 is taken from the Device Name field of the MIDI Devices LIST in the Main Menu\MIDI page.
The Devices list allows MIDI Channels to be mapped to a text name for easier identification of the devices being selected.
Note that where the Message Type being selected has a Device ID rather than a MIDI Channel (which will be the case only if the MSG Type is set to MMC Locate, SysEx or Go to Cue – see next page) then the Device ID will be displayed instead of the channel.

{FIRE} key: When pressed this transmits the displayed MIDI event for test purposes during equipment setup. Only the currently-selected event from the list of 20 is transmitted.

SCROLL CUE LIST encoder: Allows the Cue list to be scrolled whilst the EDIT MIDI page is open, allowing the Events Setup to be compared between Cues.

Default settings stored in current Show are (for each of the 20 Events):

On key = OFF
Channel = 'No device'
Message Type= blank
Value 1&2 = blank

CUE LIST PAGE - MIDI Field Touched & Event List Open

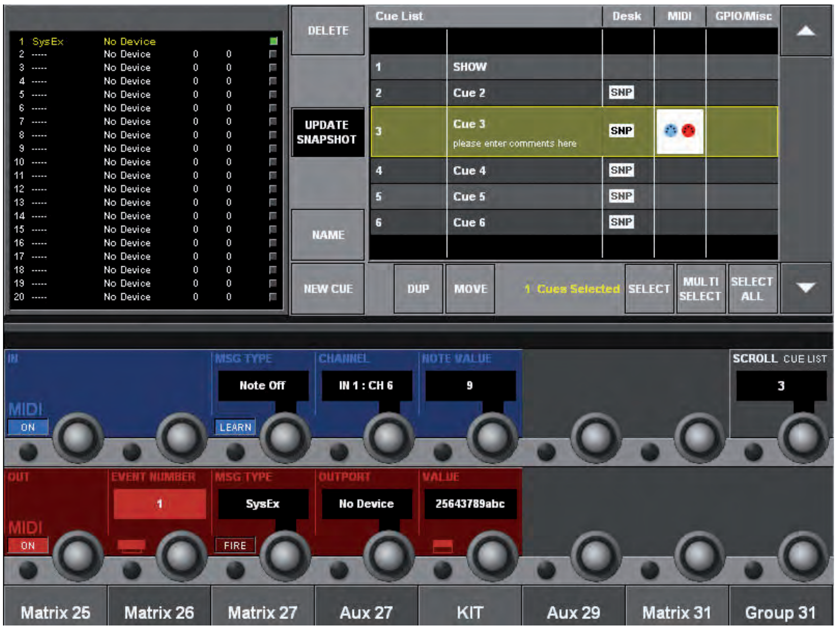


Figure 15-16: MIDI Field Touched & Event List Open.

Pressing the sub-page key in the EVENT NUMBER VST field opens the Events sub-page, which is displayed to the left of the Cue List.

Note that the events will be transmitted in the order that they appear in the list (i.e. No.1 first), although the speed of transmission of all 20 events is almost simultaneous. However the order can be important: if, for example, you wish to locate a playback machine to a timecode value and then start it playing the Locate command must be placed before the Play command.
The page gives visibility of all of the MIDI out events that have been set up for the currently selected Cue (up to 20 events).

The {SCROLL CUE LIST} encoder can be used to scroll through the Cue List whilst the Events sub-page is open. In this way it is easy to compare the MIDI Event setups for different Cues.

If more than 20 Events are necessary, the Sequencer function (see page 15-13) can be used to ‘chain’ two or more Cues together from a single Recall command.

The sub-page is closed again by pressing the sub-page key (there is no Exit button).

CUE LIST PAGE - GPIO/Misc Field Touched

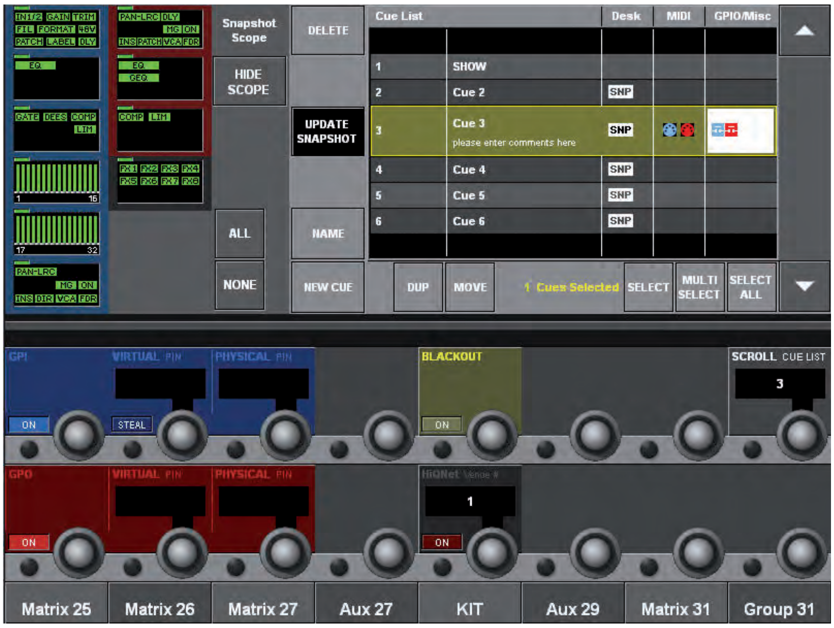


Figure 15-17: GPIO/Misc Field Touched.

GPI {ON} key: Enables/disables reception of the defined GPI signal from recalling the current Cue.

VIRTUAL PIN encoder: Allows a Cue List Virtual GPI Pin to be selected as the trigger source. The Virtual Pin must be assigned to a real (Physical) Pin in the GPIO Page. There are 80 Virtual GPI Pins available. (allows for 32 local plus up to 48 on 6 stageboxes). The Virtual pin number display is in grey text if that pin is already used by another Cue.

{STEAL} key: Only displayed if the selected Virtual pin is already assigned to another Cue.
Pressing the STEAL button reassigns the Virtual pin to the current Cue (no warning dialogue given).

PHYSICAL PIN field: displays the physical Pin number that is currently assigned to the selected Virtual Pin. (Display only). If a physical pin has not been assigned, this field will be empty.

GPO {ON} key, as above, but for outgoing GPO events. The Virtual GPO pins must be set up in the GPIO page.
There is no need for a Steal key on GPOs, several Cues can be assigned to the same v-pin. Note that although there is only one v-pin per Cue, a v-pin can be assigned to any number of physical pins in the GPIO setup page.

BLACKOUT {ON} key: Enables/disables the Blackout event on this Cue. This is intended for use in a dark scene in a theatre production.
The Blackout event switches off all console LEDs, screens and desk illumination, except for the NEXT and LAST buttons and the [F6] key. Pressing the [F6] key switches all illumination back on, but does not activate any other function assigned to it. When the illumination is back on, [F6] is returned to its normal function (if any).

HiQnet {ON} key: enables/disables the transmission of the assigned HiQnet Venue Change message upon recall of this Cue.

HiQnet encoder: Allows selection of the Venue Change number that will be transmitted from the HiQnet Ethernet port when this Cue is recalled.
Note that the IP address of the console must be set up in order to use the HiQnet functionality. These parameters are available in the Menu\system\HiQnet page (see chapter 14).

Global Filter

ISO Key Functionality

[ISO] keys have 3 states: 2 ON states and 1 OFF state.
Pressing the key will cycle round the states in the order OFF, ON1, ON2, OFF, ON1 etc.
Note that state ON2 will only exist if a partial Isolation has been set up on the strip (using the press & hold functionality – see below).

OFF state: (key not illuminated): No Isolation of any parameters.
ON1 state: (key illuminated): Full channel Isolation – purple border around channel/bus
ON2 state: (key illuminated): Partial Isolation activated on strip, if this has been set up using one of two methods : Press & Hold or Edit Global Filter mode. Indicated by the selected individual parameters displayed in purple.

Press & Hold ISO Key Functionality

To make a partial isolation: hold down an [ISO] key and make a short press on the VST function block to select the whole block. A wide purple LED-style indicator on the screen shows the isolated block state.

If the function block is NOT already zoomed, it is possible to hold down the [ISO] key and then press & hold the function block for 2 secs, in order to activate the zoom mode.

If the function block is already zoomed, individual parameters can be touched, (or the adjacent VST key pressed) and the parameter label text will change to purple to indicate selection.
The colour of that parameter in the function block touch field will change to purple to indicate parameter-level isolation.

Global Filter ON/OFF switch

A set of master controls for the Global Filter is located in the lower left-hand side of the VST area in the Cue List screen. See Fig 15-18.

Figure 15-18: Global Filter Controls

The Global Filter master {ON} key allows the complete Global Filter settings on all channels to be temporarily switched off if required.
The Global Filter Master {ON} key switches ON as soon as any ISO buttons on the console are pressed
If Global Filter On is then switched OFF, the isolation state and all indications of it on the surface will be removed (including illuminated [ISO] keys and purple Vistonics graphics). The state of these will be held in memory however, enabling it to be switched back on again later.

If Global Filter On is switched ON again, without pressing any [ISO] keys in the meantime, then the state of the filter will be restored from memory to the surface.

Clearing The Global Filter

If the Global Filter {On} key is OFF, pressing any local [ISO] key (either short or long press) will clear any previously set Global Filter, which may be being held in memory, and start a 'new' Global Filter on the surface with the newly-selected parameters.

The Global Filter On switch in the Cue List page will automatically change to the ON state when the first parameter is selected.

Global Filter settings are indicated on the Vistonics screens by means of the colour purple, as follows:

A complete* purple border around an input channel or output bus = full channel/bus isolated
A wide purple LED indicator in the top left of a function block = complete function block isolated
A narrow purple LED indicator in the top left of a function block & individual purple parameter icons in the function block = some parameters isolated
Purple parameter name(s) within a zoomed function block = parameter isolated
A horizontal purple bar across the screens (except the control bay) indicates that the whole console is in Edit Global Filter mode (see later).

*Note: if there is no FX assigned to an output bus the purple border will not encompass the FX function block.

Remember that the Global Filter of the console can also be edited directly on the console surface, without losing control of the majority of the surface, by using the [ISO] keys, as described on the previous page.

In conjunction with the Gang function, the [ISO] keys enables horizontal groups of channels, function blocks or individual parameters to be quickly set in and out of isolate (Global Filter) mode.

Edit Global Filter Mode

The {EDIT} GLOBAL FILTER key (see Fig 15-18) will switch the whole console into an edit mode which is the equivalent of 'Press & Hold on all ISO keys at the same time'. (ISO keys continue to work, but only with their 'Isolate All on this Channel' function, in this edit mode. There is no actual press & hold ISO functionality in this mode – it is not required).

In this mode the horizontal bars across all VST screens (except the control bay) will change from their normal colour (Blue or Red) to Purple, to indicate that audio can no longer be controlled from the VST screens.

Selecting function blocks or individual parameters is then done as follows: touch the function block with a short press to select the whole block (indicated by the wide purple indicator) or press & hold the function block to enter zoom mode, where individual parameters can be touch selected (or by VST key).

Note that the filter parameters being edited here are the same ones that are set by the "holding down [ISO] key" method previously described.

The reason for having two methods of editing the same parameters (locally using ISO or globally using Edit Global Filter mode) is that whilst the ISO method allows very fast, immediate control of channel filtering, even using Gang in addition to quickly set filters across the whole console, if there are a lot of parameters to be set into Isolate mode, the latching Edit Global Mode will be easier to use.

The [SEL] key is used to select whole channel to isolate mode. The [ISO] key can still also be used.

LOAD ISO WITH SHOW

The {YES}/{NO} key (its legend toggles) enables the user to recall (or not) the new Global Filter settings when a new Show is loaded.
The purpose of this feature is to allow ISO buttons to be used to protect sections of the console (e.g.. output section and master outputs) from changing when a new show is loaded in. This can be useful in a multi-band situation where the desk outputs are set up for the PA system but a visiting engineer wants to load their own show without changing the output section (or the interval music CD player for example). Setting LOAD ISO WITH SHOW to NO, and then switching the required parts of the show to ISO, will allow the show to be loaded without changing the isolated parts. The state of the key is not saved but defaults to YES when the desk is powered up.

MANAGING SHOWS

Loading Shows, copying Shows to and from USB data storage devices, and creating new Shows is done from the following page. It is reached by pressing [MENU] and then the <SHOW> tab at the top of the page (or the Show name at the top of the Cue List display area, which is at the top right corner of the main screen, can be touched).

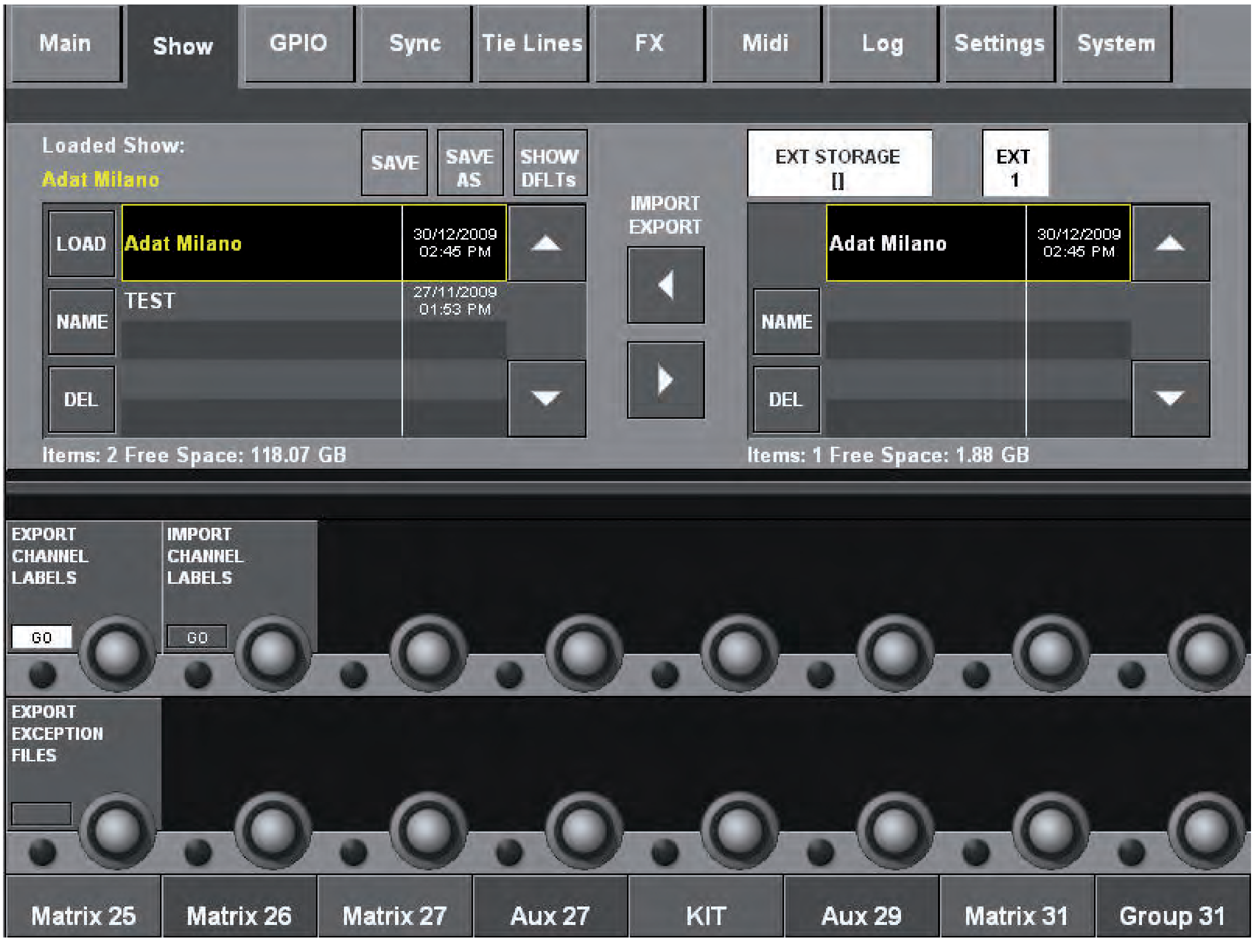


Figure 15-19: The Show Page

Note that the Show page appears as shown in Figure 15-19 if a USB data storage device is present in the USB slot. If there isn't any external memory, the right-hand side of the screen is blank, and the Export/ Import Controls are not shown. Up to 3 external storage devices can be connected: if more than one is detected, additional buttons will appear to the right of the <EXT1> button, and pressing the required button will select that device.

Flash Drive

The left-hand side of the page displays the shows which are present on the console's flash drive. The up and down-arrows on the touch screen are used to scroll though the available Show titles. Pressing and holding the arrow keys scrolls at a higher speed. Once the required show has been highlighted, in the double-height yellow-outlined box, the three buttons on the left can be used as follows:

The <LOAD> button will load the selected Show into the console (note that the currently-loaded show's title is shown in yellow text above the list).

The <NAME> button allows the user to name/rename the show, the on-screen keyboard is displayed.

The<DEL> button deletes the show from the flash drive (note that the currently-loaded show cannot be deleted). A confirmation dialogue is displayed.

Default Shows

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> button, nor renamed. The Default Shows are designed to reset the whole console back to a 'flat' starting point.

Creating A New Show

The user can select a default show or another existing show and use the <SAVE AS> button. The user will then be asked to enter a name for the new Show using the on-screen keyboard. This new show can then be edited 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 Cues. This is because of the write-protection assigned to the default Shows. It is good practice to create your new show using the SAVE AS facility before you start to make your setup on the console.

Updating A Show

A loaded show can be updated at any time by pressing the <SAVE> button. If you are not using Cues to store the state of the desk, it is essential that you save your Show in order to keep any changes made to the surface since you created the Show.

USB Data Storage Device

The right-hand side of the page displays the shows which are present on the installed USB data storage device. Normally the front panel USB Data Socket will be used (EXT1), but it is possible to connect USB data storage devices to the rear panel USB sockets (EXT2 and EXT3), and select the required device by using the <EXT1><EXT2><EXT3> buttons. The up and down-arrows on the touch screen are used to scroll though the available Show titles. Once the required show has been highlighted, in the double-height yellow-outlined box, the two buttons on the left can be used as follows:

The <REN> button allows the user to rename the show, the on-screen keyboard is displayed.
The<DEL> button deletes the show from the USB data storage device.

Exporting A Show To A USB Data Storage Device.

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

Importing A Show From A USB Data Storage Device.

Pressing the left-facing arrow will import the currently-selected Show from an installed USB data storage device. Note that the date and time of the latest save is shown in both lists.
With Importing and Exporting, a dialogue box appears showing progress. To avoid data loss do not remove the storage device until the progress box has disappeared

Export/Import Channel Labels

These keys allow the user to export a list of current Channel names in CSV (comma-separated values) format. This file can be edited in a PC spreadsheet package and then imported back into the desk. The file name is Soundcraft Vi channel labels.csv. Pressing the appropriate GO key immediately exports or imports the file to a USB memory device. These keys are only available if a USB memory device is connected. Note that there is no progress dialogue because the operation is almost instantaneous.

Export Exception Files
If the console's on-board computer malfunctions an exception file is automatically written to the flash drive. Pressing the GO key will write this file to a connected USB memory device. A service engineer may want to look at this file. This key is only available if a USB memory device is connected.

RECORDED DATA
As was mentioned previously, some settings are recorded as part of a Show, others as part of a Snapshot, and a few are not recorded at all. The following diagram (Figure 15-20) shows how this is done.

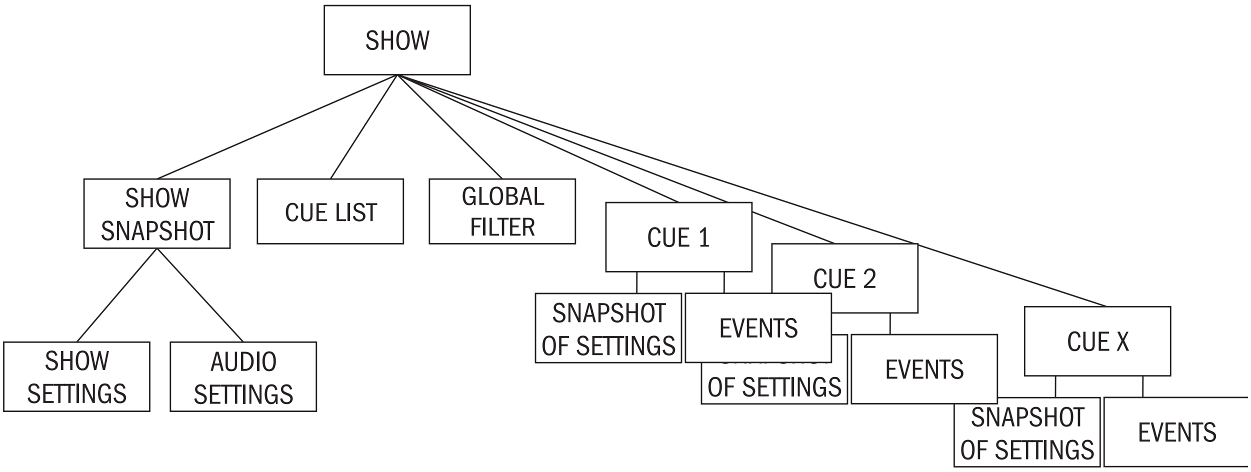


Figure 15-20: Recorded Data Structure

Note that a Show has one hidden 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 which 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. The Show Snapshot allows the complete status of the console to be recorded with the Show, even if no Cues 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; Sample Clock Settings.
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 (but only when Shows have been saved on the same console type: Vi4 or Vi6).
MIDI Configuration Setup: MIDI Channel names.
ISO switch status (Input & Output). This depends on the setting of the LOAD ISO WITH SHOW parameter in the Cue List page.
O/P Vistonics Lock Mtr switch status.
O/P Vistonics Solo/OnOff/TB switches.

All parameters in the menu/settings page.
Follow Solo switches.
Mute Group Master switch status.

Settings Recorded Within Audio Settings
This applies to the Show Snapshot and to any standard Cues.

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 settings.
All I/O controls.
*A ‘channel’ is an input, output or Master LRC channel.

Settings Not Recorded
PFL/SOLO switch status.
LRC Sel switch status.
All round Setup switch status.
User Defined switches (O/P fdr pnl) status.
All switches in Snapshot Control section.
Copy/Paste/Undo switch status.
Set Pre/Post modes switch status.
Pan/Level toggle switch status.
Upper & Lower Encoder Row Assign switches status.
Gang Mode switch status.
Solo Clear switch status.

Settings Restored To Their State At the Last Power-Down
Which Show was loaded.
LOAD ISO WITH SHOW switch status.

Show Compatibility
Vi2, Vi4 and Vi6 Consoles
From software V4.7, all shows are compatible between all Vi2, Vi4 and Vi6 consoles. running V4.7.

Previous to V4.7, shows saved ony Vi console (including Vi1) can be loaded on any Vi2/4/6. All settings will be recalled except for the channel pairing, which is not imported due to the differing vertical arrangement of channels in the fader pages.

When using files from a Vi6 on a Vi4 configured for 64 inputs or less, or a Vi1, some data in user layers may be lost because of the lower I/O configuration.

V4.0 and Older Software

There is full backward and forward compatibility between Shows saved on Version 4.0 consoles and older version consoles.

> Shows saved on V2.x software can be loaded into a V4.7 console.
>
> Shows saved on a V4.7 console can be loaded into a desk running V2.x software (without any of the features such as snapshot scope, partial isolation and events that were not present in the V2.x software).

This is achieved by converting the old Shows to V4.7 format when they are loaded on the V4.7 desk, and also saving the Show and Snapshot files in both old and new sessions when a Show is saved on V4.7.

Note: there is a maximum limit of 100 Cues when importing a V2.X Show into a V4.7 desk. If you have an existing Show with more than 100 Cues that you need to load onto a V4.7 desk, contact Soundcraft for advice about splitting the Show into two or more Shows, importing them separately, exporting them and re-combining them on an external laptop.

CONFIGURATION

In the GPIO (General Purpose Input Output) Page you can configure all GPIO channels that are available in the Soundcraft Vi Series™. To access the GPIO page, press the [MENU] key, this opens the Main menu page on the master section screen, then touch the <GPIO> tab (see also chapter 14).

Figure 16-1: GPIO Page.

The scrollable tables show the configuration of the GPIO channels. The configuration is done via the VST fields. The selections made via the VST keys and encoders are reflected in the tables on the screen. For each GPIO channel, the polarity, time (outputs only) and edge can be configured, to make it easy to interface directly to different device types.

Screen Touch-Pads

<LOCAL RACK>

Selects the GPIO in the Local Rack (16 GPIO channels).

<STAGE BOX>

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

Up and Down Arrows

The pair of Up and Down Arrows are used to scroll though the two lists. The currently-selected input and output channels are outlined in yellow. GP Inputs are displayed in blue, and GP Outputs are displayed in red.

GPI VST Keys & Encoders

GPI Field
The {ON} key enables the selected GPI function. GPIO Inputs are via opto-isolators.

Function Field
This field displays the Input function. Its encoder selects the function.

Parameter Field
This field displays the Parameter (e.g. Channel number). There is no parameter for TB INPUT. The encoder adjusts the Parameter.

Polarity Field
This field displays 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.

Edge Field
This field displays the triggered Edge. The encoder adjusts the triggered edge between rising/falling/both.

GPO VST Keys & Encoders

GPO Field
The {ON} key enables the GPO function. GPIO Outputs are via pairs of relay contacts.

Function Field
This field displays the Output function. Its encoder selects the function.

Parameter Field
This field displays the Parameter (e.g. Channel number). There is no parameter for TB OUT. The encoder adjusts the Parameter.

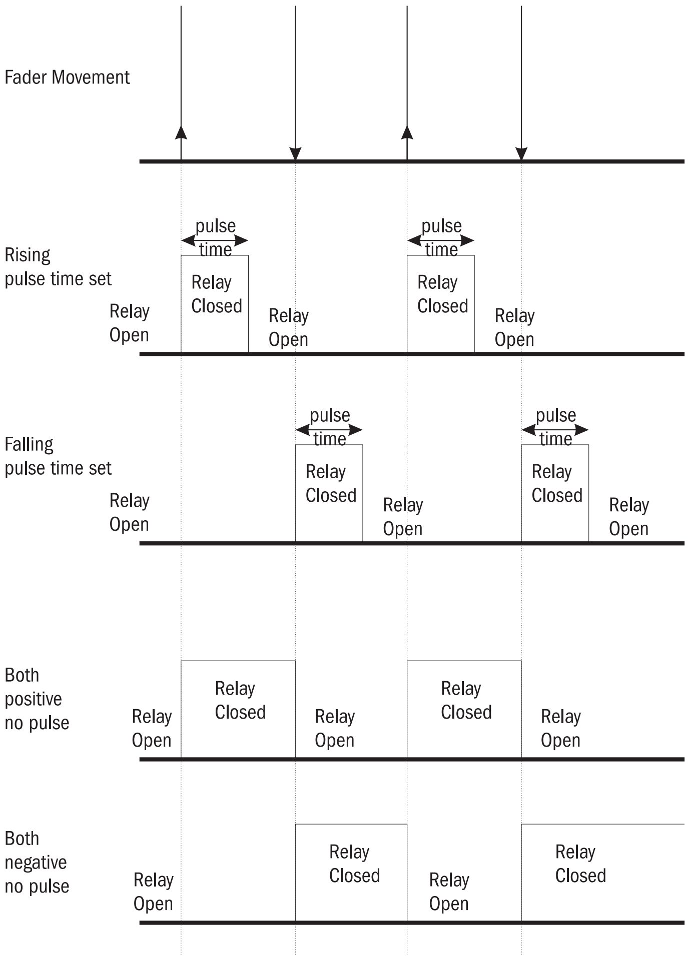
Polarity Field
This field displays the Polarity of the Output. The encoder changes the polarity between positive (+) and negative (-). This field is only available if the {EDGE} field is set to BOTH.

Time Field
This field displays 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, i.e. the relay contacts stay in their new position).
It is suggested that a pulse time is set only when the EDGE field (see below) is set to 'RISING' or 'FALLING. It is also suggested that no pulse time is set when the EDGE field is set to 'BOTH'. See Figure 16-3 for a timing diagram of relay operation using fader start.

Edge Field
This field displays the triggered Edge. The encoder adjust the triggered Edge between rising/falling/both.

	FUNCTION	PARAMETER	POLARITY	TIME	EDGE
INPUTS	CH MUTE	1-64 (channel)	POSITIVE NEGATIVE		BOTH RISING FALLING
INPUTS	F KEY LED	1-6 (f key)	POSITIVE NEGATIVE		BOTH RISING FALLING
INPUTS	TB INPUT		POSITIVE NEGATIVE		BOTH RISING FALLING
INPUTS	DIM MON	A or B (monitor)	POSITIVE NEGATIVE		BOTH RISING FALLING
OUTPUTS	FDR START	1-64 (channel)	POSITIVE NEGATIVE	0-500mS 0-500mS 0-500mS 0-500mS	BOTH RISING FALLING
OUTPUTS	F KEY	1-6 (f key)	POSITIVE NEGATIVE	0-500mS 0-500mS 0-500mS 0-500mS	BOTH RISING FALLING
OUTPUTS	TB OUT		POSITIVE NEGATIVE	0-500mS 0-500mS 0-500mS 0-500mS	BOTH RISING FALLING

Figure 16-2: Summary Of Available Settings.



Set a pulse time
when using
'RISING' or 'FALLING'.

Do not set a pulse time
when using 'BOTH'.

Figure 16-3: Relay timing diagram.

HARDWARE
Schematic Diagram

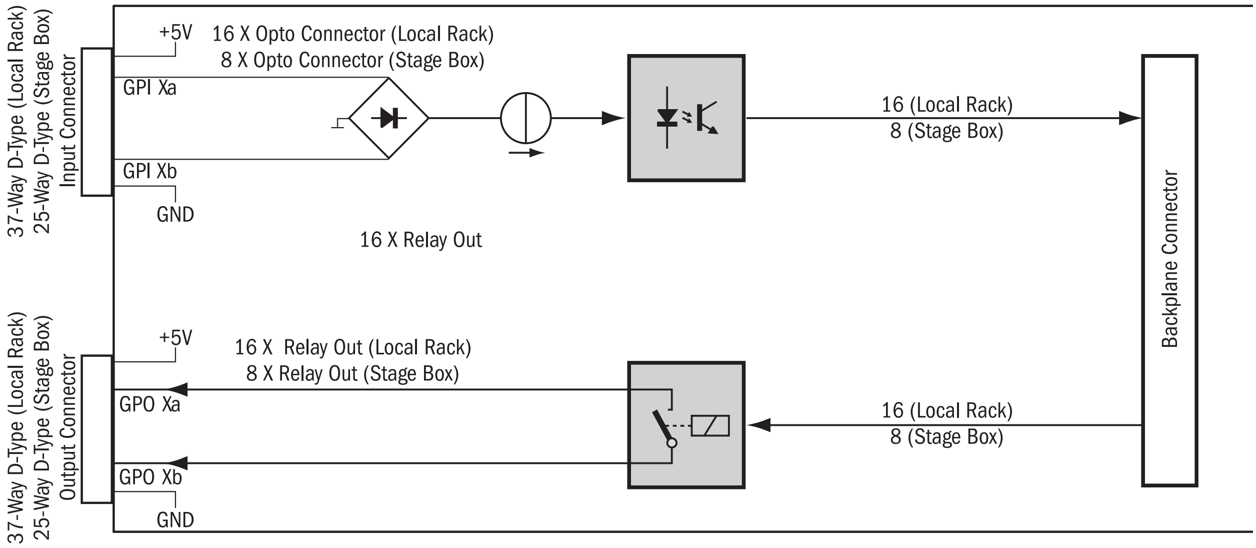


Figure 16-4: Schematic Diagram.

For general-purpose applications requiring total electrical isolation, the GPIO card 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).

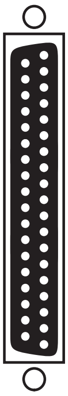
Inputs
Control inputs (GPI Xa/b) are completely independent and electrically isolated. They may be used either with the internal +5 VDC supply voltage, or with external voltages of 5 to 24 VDC, regardless of the polarity. Total current supplied by all +5 VDC pins of one card must not exceed 600 mA.

Outputs
Control outputs (GPO Xa/b) are completely independent, electrically-isolated relay contacts, closed if active. Contact rating is 0.5 A for 125 VAC, 1 A for 30 VDC, or 0.3 A for 110 VDC. The +5 VDC supply voltage or the ground (GND) terminals, together with the relay contacts, may be used to generate an output signal. Total current supplied by all +5 VDC pins of one card must not exceed 600 mA.

Pin Lists

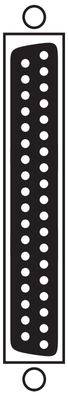
LOCAL RACK

INPUTS



Pin	Signal	Pin	Signal
		19	GND (0V)
37	VCC (+5V)	18	GND (0V)
36	VCC (+5V)	17	GND (0V)
35	GPI 16b	16	GPI 16a
34	GPI 15b	15	GPI 15a
33	GPI 14b	14	GPI 14a
32	GPI 13b	13	GPI 13a
31	GPI 12b	12	GPI 12a
30	GPI 11b	11	GPI 11a
29	GPI 10b	10	GPI 10a
28	GPI 9b	9	GPI 9a
27	GPI 8b	8	GPI 8a
26	GPI 7b	7	GPI 7a
25	GPI 6b	6	GPI 6a
24	GPI 5b	5	GPI 5a
23	GPI 4b	4	GPI 4a
22	GPI 3b	3	GPI 3a
21	GPI 2b	2	GPI 2a
20	GPI 1b	1	GPI 1a

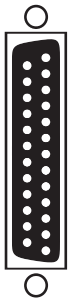
OUTPUTS



Pin	Signal	Pin	Signal
		19	GND (0V)
37	VCC (+5V)	18	GND (0V)
36	VCC (+5V)	17	GND (0V)
35	GPO 16b	16	GPO 16a
34	GPO 15b	15	GPO 15a
33	GPO 14b	14	GPO 14a
32	GPO 13b	13	GPO 13a
31	GPO 12b	12	GPO 12a
30	GPO 11b	11	GPO 11a
29	GPO 10b	10	GPO 10a
28	GPO 9b	9	GPO 9a
27	GPO 8b	8	GPO 8a
26	GPO 7b	7	GPO 7a
25	GPO 6b	6	GPO 6a
24	GPO 5b	5	GPO 5a
23	GPO 4b	4	GPO 4a
22	GPO 3b	3	GPO 3a
21	GPO 2b	2	GPO 2a
20	GPO 1b	1	GPO 1a

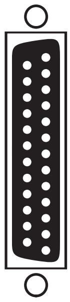
STAGE BOX

INPUTS



Pin	Signal	Pin	Signal
		13	GND (0V)
25	VCC (+5V)	12	GND (0V)
24	VCC (+5V)	11	GND (0V)
23	VCC (+5V)	10	GND (0V)
22	VCC (+5V)	9	GND (0V)
21	GPI 8b	8	GPI 8a
20	GPI 7b	7	GPI 7a
19	GPI 6b	6	GPI 6a
18	GPI 5b	5	GPI 5a
17	GPI 4b	4	GPI 4a
16	GPI 3b	3	GPI 3a
15	GPI 2b	2	GPI 2a
14	GPI 1b	1	GPI 1a

OUTPUTS



Pin	Signal	Pin	Signal
		13	GND (0V)
25	VCC (+5V)	12	GND (0V)
24	VCC (+5V)	11	GND (0V)
23	VCC (+5V)	10	GND (0V)
22	VCC (+5V)	9	GND (0V)
21	GPO 8b	8	GPO 8a
20	GPO 7b	7	GPO 7a
19	GPO 6b	6	GPO 6a
18	GPO 5b	5	GPO 5a
17	GPO 4b	4	GPO 4a
16	GPO 3b	3	GPO 3a
15	GPO 2b	2	GPO 2a
14	GPO 1b	1	GPO 1a

Figure 16-5: Pin Lists.

GENERAL

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 the Soundcraft 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.

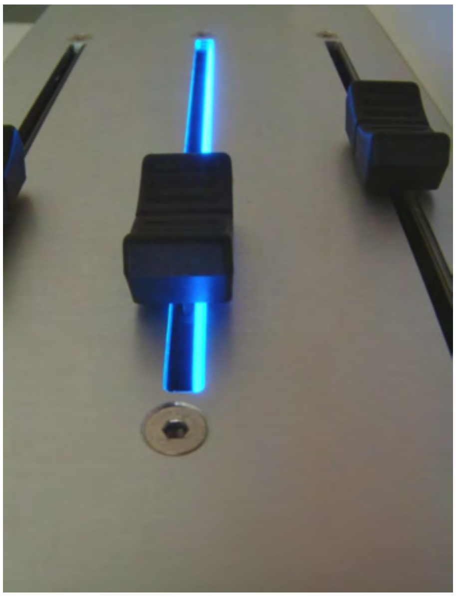


Figure 17-1: Soundcraft FaderGlow™.

Colour Code

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

INTRODUCTION

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 last paste operation can be quickly reversed with an UNDO function. Processing blocks (eg EQ) or even individual parameters (eg mic gain control) within a processing block are selected for copying via touch screen selection on the same Vistonics™ II screens that are used for audio control functions.
A range of channels or busses can also be easily copied and pasted to another area of the desk.

The Library functionality allows you to select any set of parameters in use on the desk, ranging from a single channel's EQ setting, to a range of channels set up for a drum kit, to be stored in the internal library and recalled at will. The libraries can be exported to, or imported from, the USB memory stick, allowing you to build up your own portable channel and processing element libraries that can easily be transferred to any Vi console you need to work on.

This is done independently of the Show Files which allow entire desk settings to be exported.
The console's default Library includes a selection of factory default library items for a number of common sources, plus a set of 'Flat' channels and processing elements which can be used for resetting areas of the console to default state when it is not desirable to reload a complete Default Show.

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, where the surface is used for selection.
Note that in these modes control of channel strip audio from the Vistonics™ screens is not possible!

CONSOLE CONTROL KEYS

The EDIT keys on the control surface are used to initiate COPY and PASTE modes.
[Copy] mode is used for **source** selection.
[Paste] mode is used for **destination** selection.
[UNDO] can be pressed after a PASTE operation to restore the previous settings. If you wish to do this note that [UNDO] must be pressed before PASTE is switched off.

Figure 18-1: EDIT Keys

COPY & PASTE PRINCIPLES

In COPY mode, it is neccessary to select the required items. Selecting them places the items onto the clipboard. Single or multiple channels or busses can be collected, or single or multiple elements from within the same channel or bus can be collected.

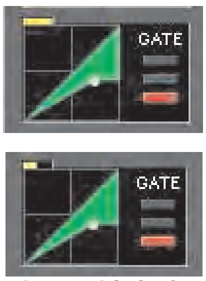
DATA SELECTION & INDICATORS

Full Channel

A full Channel can be selected in both COPY and PASTE modes by pressing the [SEL] Key.
A range of full Channels can be selected by holding the [SEL] Key of the first Channel and pressing [SEL] of the last Channel.
The grey Channel border is replaced with a yellow border to indicate a whole Channel selection.
The SEL keys illuminate in blue, in COPY or PASTE modes, to indicate their function as selects rather than as a Solo.

Figure 18-2: Selecting a Channel

Function blocks



In COPY mode, the selection of a function block can be toggled by touching the 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.

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

Figure 18-3: Selecting a Function Block

Parameters

A single Parameter can be selected by touching the corresponding Rotary encoder or by pressing the

small key to the left of the encoder. The selected state 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.
The parameter name will change to yellow text to indicate that it is selected.

Figure 18-4: Selecting a Parameter

HINT: You can enter the 'Zoom' mode where individual parameters can be selected, 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 'Zoom' mode.

COPYING CHANNEL OR BUS PARAMETERS

· Press [COPY].
[COPY] lights.
· Collect Channel Parameters by pressing [SOLO/SEL] for the whole Channel
or
Touch Fields for functional groups like EQ
or
Long press on Touch fields to enter Zoom mode for individual Parameters.
Parameters are copied to the clipboard.

· Press [PASTE].
[PASTE] lights, [COPY] will be switched off.
· Select the destination channel by pressing [SOLO/SEL]
or
touching any VST field on the channel.
The clipboard content is immediately copied to each selected channel.
· End the function by pressing [PASTE] again.

The same procedure can be used for busses: you must be in the ALL BUSSES fader page to collect functional groups or parameters

COPYING A BUS MASTER INCLUDING ALL CHANNEL SEND LEVELS

· Press [COPY].
[COPY] lights.
· Select a Bus Master with [SEL]. This can be done either in the central output fader section (pages A-D), or the ALL BUSSES page.
· Press [PASTE].
[PASTE] lights, [COPY] will be switched off.
· Activate “include send levels” in the central page (choose YES).
· Select the destination Bus with [SEL].
· End the function by pressing [PASTE].

HINT: Bus copy can be performed between different Bus Types (AUX<>GRP, GRP<>AUX). In this case the ON/OFF state of the AUX sends will be equivalent to Group Routing ON/OFF.

PARAMETERS NOT INCLUDED IN CHANNEL & BUS COPY MODES

When you copy a whole channel, a whole bus, or the input and output blocks within the channel or bus, not all parameters are copied. For example if a whole channel is copied, the Input patch, gain, Insert and direct out settings are NOT copied.

The sections below list the various parameters that are NOT included if the whole channel or bus is selected for Copy using the SEL button.

Note that additional parameters can be added to –or subtracted from- the channel copy by using the ‘long-press’ on the touch screen in the required field and then selecting or deselecting various parameters.
It is never possible to select the Insert point or Direct Out however.

CHANNEL COPY: ITEMS NOT INCLUDED IN COPY WITH SEL BUTTON

Input 1/2 switch
Input 1 & 2 patch
Mic Input Gain
All associated switches (48V, PAD, Phase Inv etc)
Insert Point – all aspects
Direct Output – all aspects
Fader and Mute
VCA & Mute Group Assignment
FX settings

BUS OUTPUT COPY: ITEMS NOT INCLUDED IN COPY WITH SEL BUTTON

Output patch
Insert Point – all aspects
FX settings

Note that the Aux send levels (or routing switch status for Groups) from input channels to a bus also always copied when the bus master is copied, but you can choose whether to paste the sends or not using the {INCLUDE SEND LEVELS} key in Paste mode.

EXAMPLE: COPYING A WHOLE CHANNEL, INCLUDING THE ‘IN1 PATCH’ PARAMETER

. Press [COPY] to enter Copy Mode

. Press [Solo/SEL] button on required input channel to select it to the clipboard.

. Touch the Input touch field on the channel strip with a long press (2s) to enter the ‘zoom’ mode for the input parameters. Notice that only the Trim, Filters and Delay parameter are already selected (indicated by yellow text).

. Touch the encoder (or press the adjacent Vistonics button) in the IN1 PATCH field to add the patch parameter to the copied items.
(the IN1 PATCH text changes to yellow to indicate selection)

. Press PASTE and select the destination channel(s) to paste the selected channel including the patch setting.

COPYING FX PARAMETERS

Parameter settings from any of the 8 internal Lexicon FX units can be copied from one device and pasted to another.

To make selection of the source and destination units easy, a set of 8 *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.

Alternatively, if the FX unit is assigned to a channel or bus and is therefore visible on the channel or bus strip, then the source and destination 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 and pasted for each unit – so for example if LEX1 is currently set to the Small Hall FX type, then only the Small Hall parameters will be copied, if LEX1 is selected to copy.

When the settings are pasted to another unit, the target unit will have its FX Type changed to Small Hall, and only the Small Hall parameters will be pasted. None of the other 28 FX Types within the target FX unit will have their settings changed.

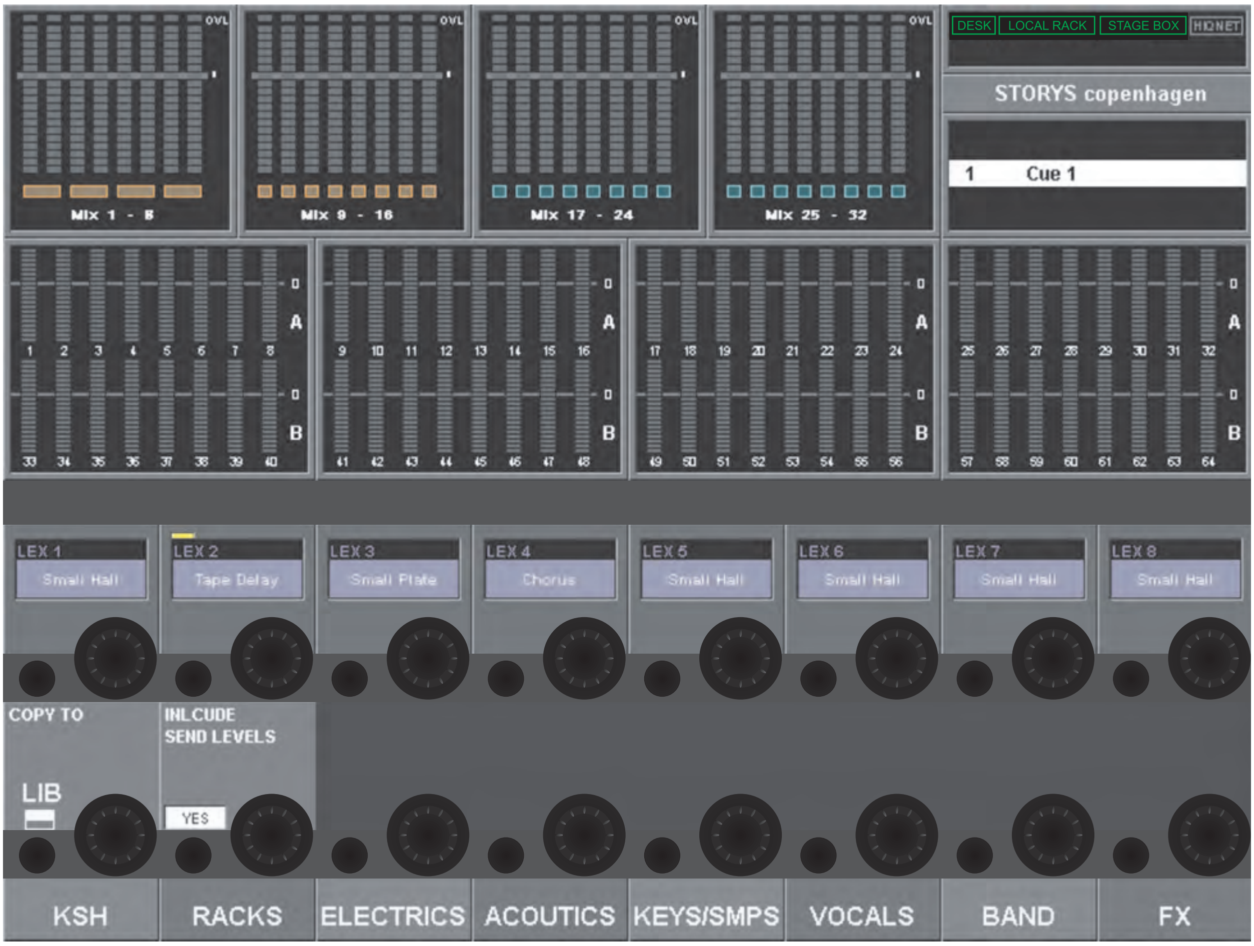


Figure 18-5: The 8 FX-select encoders in Copy mode with LEX2 selected

To copy an FX unit's settings to the clipboard:

. Press [COPY]

. Select the required FX Unit, either using the FX Select Encoders displayed in the central screen, or touching the FX unit's icon on the Channel or Bus strip, if one has been assigned and is visible on the strip.

Parameters are copied to the clipboard

. Press [PASTE]

. Select the destination FX unit by touching the required FX Select Encoder (or adjacent button) in the central screen.

or

touching any touch field on the target channel or bus strip.

or

pressing [SOLO/SEL] on the target channel or bus

T*he FX unit settings on the clipboard are immediately copied to the destination FX unit.*

. The operation can be undone by pressing [UNDO], before exiting Paste mode.

. End the function by pressing [PASTE] again.

LIBRARIES

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.
The ability to export complete Libraries or individual entries to a USB stick allows you to extract saved items from the console and later import them to another console.

The Library system 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 (eg: EQ, Dynamics, Channel, etc).
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. The folder structure is as follows:

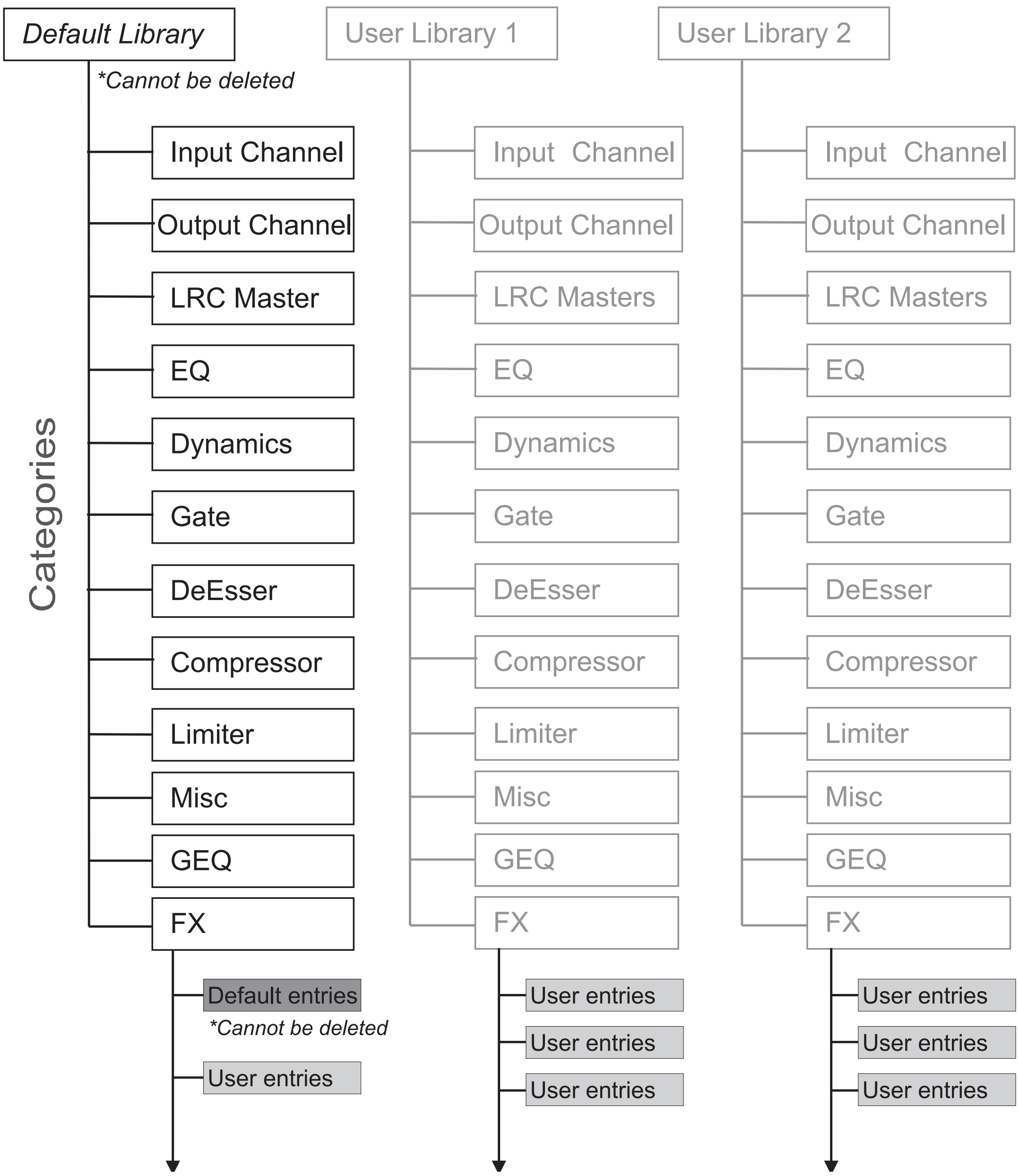


Figure 18-6: Library Folder Structure

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. The update will be in the form of a web download which can be copied to a USB stick and transferred to the console. This may be included as part of the routine software updates for 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 <100kB for individual single channel Entries).

It is recommended for easier data management that each user creates their own Library on the console, populates it with their personal settings and then exports the Library to their own USB stick.
The Library can then be imported to the next console, added to if required and then re-exported (see later section Exporting and Importing Libraries).

Categories
The list of Categories can be seen in the diagram on the previous page.
Categories in the Library structure are predefined and cannot be changed by the user.
Note that only items that have a Category can be stored to the Library – it is not possible to store every type of parameter 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 InputChannel Category.

The Misc Category
Most Categories correspond directly to the console function of the same name, but the *Misc* Category requires further explanation.
This 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 to allow storing of any of the various channel parameters which do not have their own Category, as well as any combination of these parameters.

NAVIGATING AND MANAGING LIBRARIES

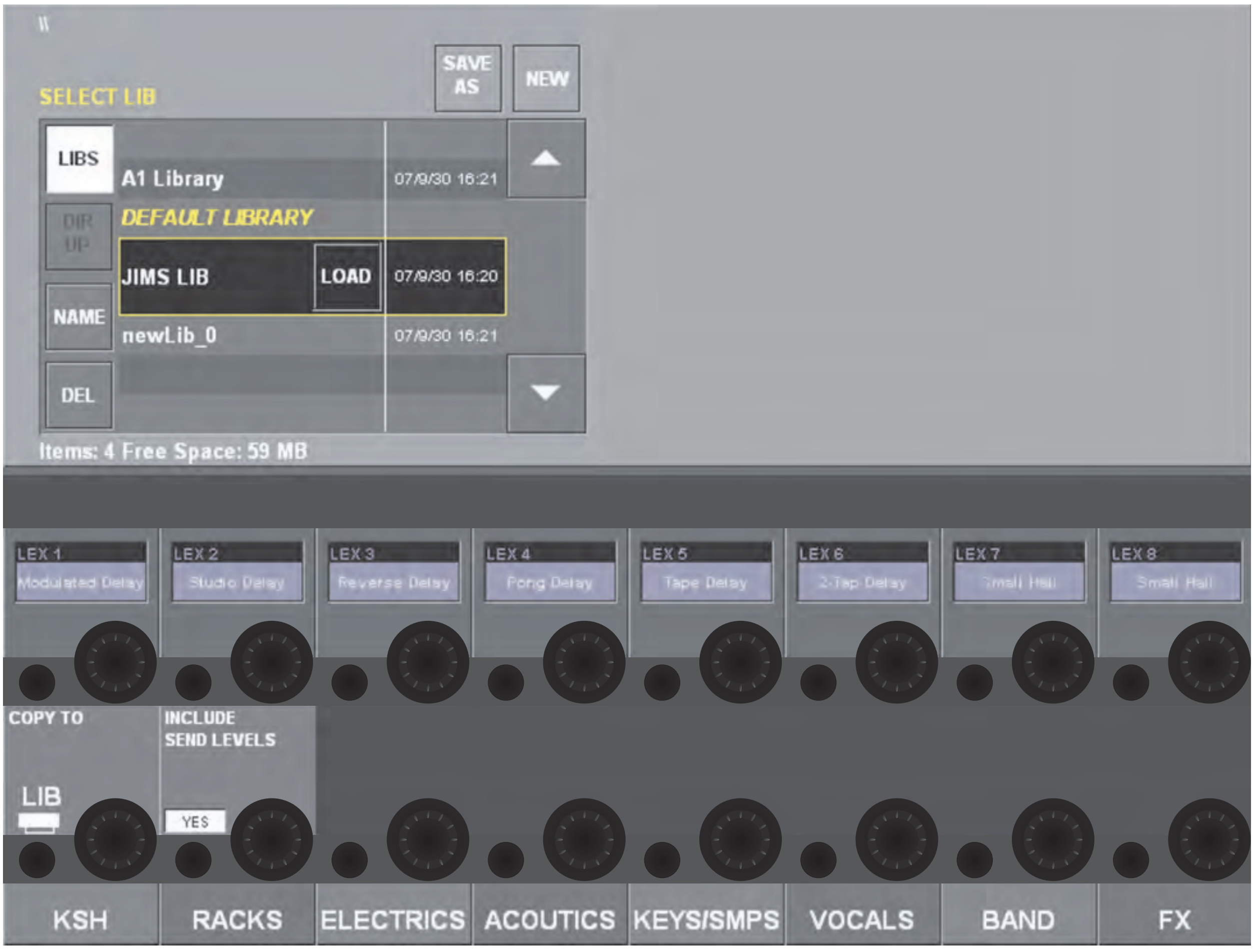


Figure 18-7: Copy to Library page

To perform operations on the Library, either the **Copy to LIB** page or the **PASTE from LIB** page must be opened. To do this, press either [COPY] or [PASTE] followed by the {LIB} key in the bottom left corner of the central Vistonics screen.

TITLE info: The text line in the top left corner of the screen, beginning \\ always shows which Library and Category is currently selected.

{LIBS} Jumps to the top of the Library folder structure, regardless of whether the library is currently displaying Category or Entry level. Pressing {LIBS} when at the top *Library* level jumps back down to the previously selected Category level.

{DIR UP} Moves up the Library folder system one step with each press. There are three levels: *Libraries* (top) \ *Categories* \ *Entries* (bottom)

{SEL} Located in the selection bar, this button is used to drill down to the next level. At the top LIBS level, the {SEL} button selects the highlighted Library to be loaded. The currently loaded Library is indicated by its name being displayed in yellow text in the list.

{NAME} Opens the QWERTY keyboard to allow the name of the currently highlighted Library or Entry to be edited. Note that *Category* names cannot be edited, so the {NAME} button is greyed out at this level.

{DEL} Deletes the currently highlighted Library or Entry. Since the Category level folders cannot be deleted, the {DEL} button changes to become a {**CLR}** button when in the Category level. Pressing {CLR} will delete all Entries within that Category. Confirmation dialogues are displayed before all Delete or Clear operations. (facing page to previous screenshot)

SAVE AS button: Only displayed at the top *Library* level.
Saves a copy of the *currently loaded* Library, with a new name.
The QWERTY keyboard is opened to allow the new name to be entered.

NEW button: Only displayed at the top Library level.
Creates a new empty Library with the default name **newLib_x**.
The new Library contains only the folder structure, but no Entries.

COPY TO LIBRARY

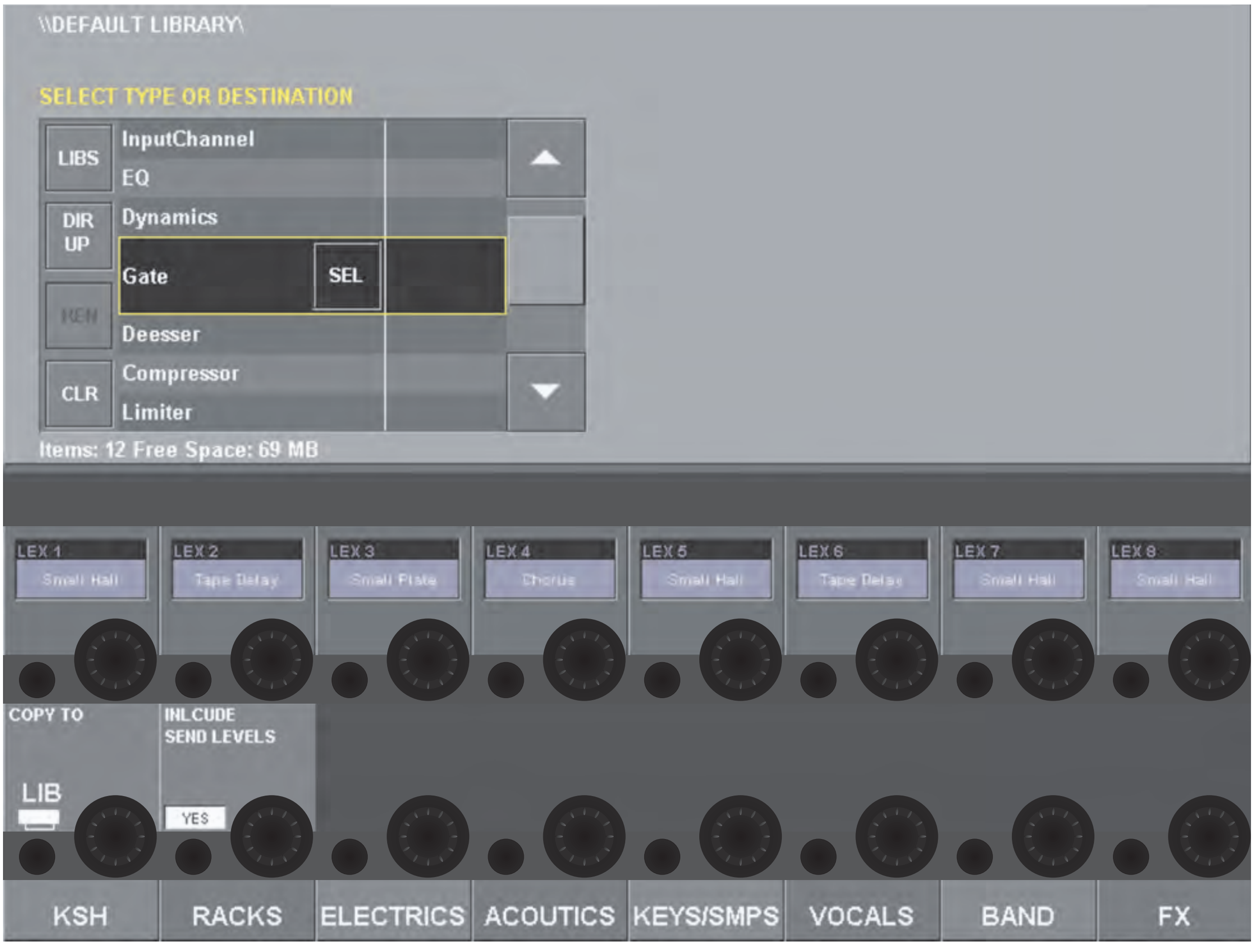


Figure 18-8: Copy to Library page

Copying an EQ to the library

- Press [COPY] on DESK
- Press {LIB} key in “COPY TO” field in the central screen
- Touch the EQ touch field on an input or output channel strip

  *The library will **automatically** change to the EQ library*
- A new library item will automatically be stored in the EQ catagory, with the name newItem_x
- Press <NAME> to open the QWERTY keyboard and rename the new item
- Press [COPY] on DESK to return the console to normal operation.

PASTE FROM LIBRARY

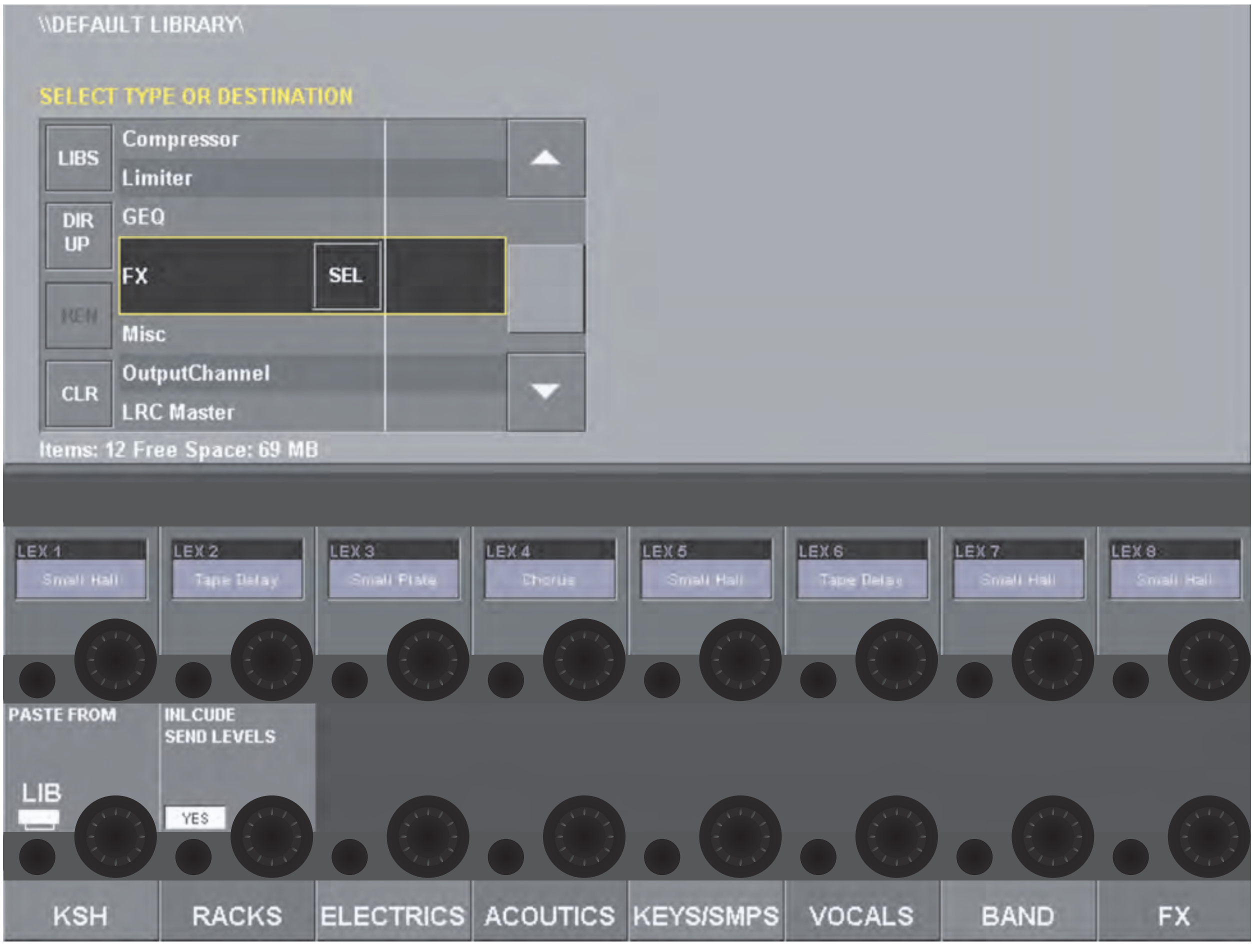


Figure 18-9: Paste from Library page

Pasting an EQ from the library

- Press [PASTE] on DESK
- Press {LIB} in "PASTE FROM" field in central screen

Method 1

- Select the desired Library /EQ and the desired library entry in the list
- Press [SEL] or the EQ Touch field on the destination Channel to load the Equaliser settings to that channel.

Note: it does not matter which touch field is touched on the destination channel – as the system already knows an EQ has been selected from the library, it only needs to know which channel you want to paste it to.

Method 2

- Touch the EQ field on an input or output channel' strip. A yellow marker will appear in the touch field border to indicate selection
  The library will automatically change to the EQ library category, and the available EQ entries will be visible
- Press <LOAD> within the library list on the central screen to load the EQ to the Desk

EXPORTING AND IMPORTING LIBRARIES

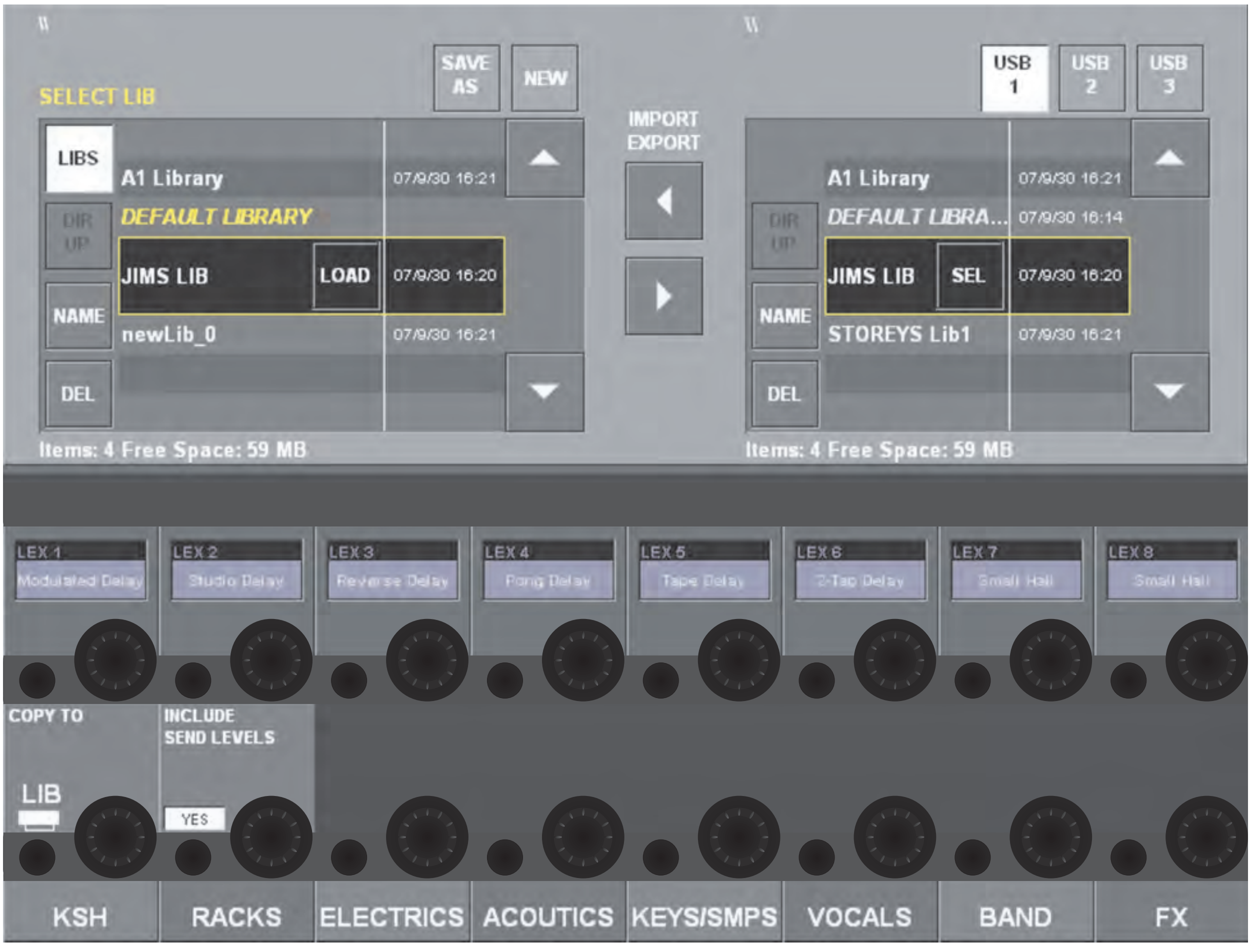


Figure 18-10: Importing to & Exporting from a USB memory device.

If a USB memory device is inserted into one of the console's USB ports, an additional window will be visible on the right-hand side of the Library page, as shown above.

This right-hand window shows the contents of the USB memory device, which will initially be empty if it has no Libraries stored on it. The {USB1-3} buttons can be used to select the required memory device if more than one storage device is connected.

Exporting

Complete Libraries

To export a Library to the USB memory device, press the {LIBS} button to move to the top of the console's internal Library structure, and use the {scroll arrow} buttons to position the cursor on the required Library in the left-hand window.

Press the {right-arrow} button between the two windows to export the selected Library to the USB memory device.

Exporting the Library creates a new folder called *Libraries* on the USB stick, and then copies the entire Library folder structure to that folder on the USB stick

The folder structure can be examined later on an external computer, and a copy made for archiving/back-up purposes.

Entries
To export an individual Entry to the USB stick, the full Library folder structure must already be present on the stick.
Using the {DIR UP} or {SEL} buttons, position the cursor on both the internal Library and the USB stick at the same level – showing Entries.
Then use the {right-arrow} button between the windows to copy the selected Entry to the USB stick.

Importing

To import a Library or Entry from the USB stick, simply use the cursor {scroll arrow} buttons to select the required Library on the USB stick in the right-hand window.
Then press the {left-arrow} button between the windows to copy the selected Library to the console.
A dialogue box indicates the progress of the copy operation.

SOUNDCRAFT Vi Series™ FEATURES AND SPECIFICATIONS

AUDIO CHANNELS

Max number of simultaneous mixing channels
Soundcraft Vi6™ : 96 mono inputs into 35 Outputs. Pairs of mono inputs can be linked to create stereo channels.
Soundcraft Vi4™ : 96 mono inputs into 35 Outputs. Pairs of mono inputs can be linked to create stereo channels.
Soundcraft Vi2™ : 96 mono inputs into 35 Outputs. Pairs of mono inputs can be linked to create stereo channels.

Insert points
24 insert send/return pairs can be configured (using available I/O) and assigned to any of the 96 inputs or 35 output channels

Direct Outputs
All input channels can have direct outputs in addition to their internal bus routing, assuming sufficient I/O is available (eg via 64ch optical MADI card, see below)

Busses
32 Grp/Aux/Matrix*, plus main LCR Mix and LR Solo busses.
* a maximum of 16 matrix outputs can be configured.

I/O CAPABILITY

The following I/O is available and can be patched to any channel input, direct output, bus output or insert point as required:

Local Rack Inputs
16 analogue line inputs
3 analogue mic/line inputs
1 Talkback Mic input (mounted on control surface – 2 parallel sockets front/rear)
8 pairs of AES/EBU inputs (=16 channels)
64ch MADI In via optical SC connectors

Local Rack Outputs
16 analogue line outputs
8 pairs of AES/EBU outputs (= 16 channels)
LCR Local monitor A analogue line outputs
LR Local Monitor B analogue line outputs
TB line output
64ch MADI Out via optical SC connectors

Stagebox Inputs
64 analogue mic/line inputs (with remote gain control, PAD, 48V and pre-A-D 80Hz HPF). This assumes Vi4 has been specified with 64 inputs on stagebox.

Stagebox Outputs
32 analogue line outputs

MISCELLANEOUS

Connection from local rack to stagebox
Standard fit: Cat 5e Neutrik Etherflex cable ZNK CT2672601.
Optional: Fibre Optical interface card with 150 or 200m cable (additional cost).

Max distance, local rack to stagebox:
80m using flexible reel-mounted Cat5 cable (Neutrik Etherflex only).
130m using Cat7 permanent installation cable (Amp Netconnect 600MHz PiMF, part no. 57893-x).
1500m using a single run of multimode 50/125 optical fibre.
600m using 3 X 200m reels of multimode 50/125 optical fibre joined in series.

GPIO facility
16 GPIO inputs and outputs on the local rack
8 GPIO inputs and outputs on the stagebox (All outputs are relay contact closure)

MIDI
1 MIDI Input and 2 MIDI Outputs on rear of control surface.

CHANNEL PROCESSING

Inputs
Analogue gain (remote control of stagebox or local mic preamp)
Digital Gain Trim (+18/-36dB)
Delay (0-100ms)
HPF, LPF (variable 20-600Hz and 1-20kHz)
4-band fully parametric EQ, shelf mode on HF/LF.
Compressor (variable threshold, attack, release, ratio, makeup gain with ‘auto’ mode)
Limiter (variable threshold, attack, release)
Gate or De-Esser. Gate switchable to ducker.
Insert point for external processing.
Pan – LR or LCR switchable.
Direct Output, patchable to any I/O and with selectable tap-off point.

Ouputs
HPF (variable 20-600Hz)
4-band fully parametric EQ, shelf mode on HF/LF.
Compressor, Limiter
Delay (0-1sec)
Insert point for external processing.
Pan (Output bus to LCR) – LR or LCR switchable.
Bus Feed feature – allows switched routing of one bus to another.

Graphic EQ 1/3-octave (with FX Card)
Assignable Lexicon Multi-FX processors x8 (with FX Card)

CONTROL SURFACE

Inputs

Soundcraft Vi6™ 32 input faders, switchable in 3 fixed layers to access 96 inputs.
Soundcraft Vi4™ 24 input faders, switchable in 3 fixed layers and meter screens to access up to 96 inputs.
Soundcraft Vi2™ 8 input faders, switchable in 3 fixed layers and meter screens to access 96 inputs.

Vistonics II channel strip interface, each Vistonics controls 8 input channels.
The Vistonics II interface contains 16 real knobs and switches and a touch screen.

Fader tray contains motorised fader, Mute, Solo, Isolate and F (user defined) switches, plus one assignable rotary encoder with LED ring. This encoder is globally assignable to Gain, Pan, Gate Threshold, or one of 2 user-definable parameters.

Input level and gain reduction meter is located above each fader.

Input faders can be assigned to the 16 VCA (control group) masters and/or 4 Mute Groups.

Input faders can be switched to control all 32 Grp/Aux/Matrix Outputs, or can control an individual Aux send mix, using the switchable 'Follow Solo' function. Soundcraft Fader Glow™ clearly indicates using colours when faders are not controlling inputs.

Outputs

8 assignable Output faders, plus 2 dedicated LR and C Master faders, plus 16 assignable rotary Output faders. Output faders are colour-coded using Soundcraft Fader Glow.

Output faders can be assigned to the 16 VCA (control group) masters and/or 4 Mute Groups.

Single Vistonics II interface for Output processing control, also functions as complete meter overview display for all Inputs & Outputs, plus snapshot Cue List and diagnostics info display.

Misc

Gang mode for temporary linking of any number of channels for quick adjustment and setup
Controls for Mute Group and VCA Group assignment.
Controls for assignment of Vistonics rows to bus sends (when channel parameters are not selected to Vistonics).
Snapshot automation controls
Talkback & Oscillator controls
Controls for Monitor Output level, phones level and Solo Trim and blend level.

Vi Series TYPICAL 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	
22Hz-22kHz	
Stagebox Mic In (min gain) to Local Line Out	<0.003% @ 1kHz
Stagebox Mic In (min gain) to Local Line Out	<0.020% @ 1kHz
Local Line In to Line Out	<0.003% @ 1kHz
Mic Input E.I.N.	<-126dBu (150W source)
22Hz-22kHz bandwidth, unweighted	
Residual Noise	-95dBu
Stagebox line output; no inputs routed, Mix fader @0dB	
CMRR	80dB @ 1kHz
Stagebox Mic input	
Sampling Frequency	44.1kHz, 48kHz
Latency	
Stagebox Mic Input to Local Line output	< 2ms @48kHz
AES/EBU Input Sample Rate	32–108kHz (with SRC enabled)
DSP resolution	40-bit floating point
Internal clock	
Accuracy	< +/-50ppm
Jitter	< +/-5ns
External Sync	BNC Wordclock, AES/EBU sync in, Video sync in
Input & Output Levels	
Mic Inputs	+28dBu max
Line Inputs	+18dBu max
Line Outputs	+18dBu max
Nominal Operating Level	0dBu (-18dBFS)
Input & Output Impedances	
Mic Inputs	2k7Ω
All other analogue Inputs	>10kΩ
Line Ouptuts	<75Ω
AES/EBU Outputs	110Ω
Oscillator	20Hz to 20kHz/Pink/White Noise, variable level

Stagebox HP Filter	80Hz fixed, 12dB per octave
Channel HP filter	20Hz-600Hz, 18dB per octave
Channel LP filter	1kHz-20kHz, 18dB per octave
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= 0.3-8.7or shelving
Metering	Internal 20-segment LED bargraphs plus 9-segment gain reduction meters for all inputs and Outputs. Peak hold variable from 0-2s.
Mains Voltage operating range	90-264V, 47-63Hz, autoranging
Mains Power Consumption	
Control Surface:	155W (165W redundant option)
Local Rack:	140W (150W redundant option)
Stagebox:	140W (150W redundant option)
Weights (without flightcases)	
Control Surface (Vi6)	63kg (140lb)
Control Surface (Vi4)	53kg (117lb)
Control Surface (Vi2)	26kg (57lb)
Local Rack	25kg (55lb)
Stagebox	16kg (35lb)
Operating Temperature Range	0°C - 45°C (32°F - 113°F)
Relative Humidity	0% - 90%, non-condensing T_a = 40°C (104°F)
Storage Temperature Range	-20°C - 60°C (-4°F - 140°F)

Soundcraft reserves the right to make changes to the above data without prior notice. E&OE.

General

The Vi Processor™ Card fitted to the Soundcraft Vi Series™ contains 8 powerful LEXICON® Effects Processing Units and 35 high-quality BSS® 30-band Graphic Equalisers.

LEXICON® Effects

Each Effects 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, fed from an Aux send.
Each FX Unit supports up to 30 different professional LEXICON® Effects.

Effect Parameters can be easily changed via the VST Screens at a location on the Surface corresponding to where the FX is inserted or patched. Additionally, the Parameters can be viewed and changed in the FX Overview Page in the main menu.

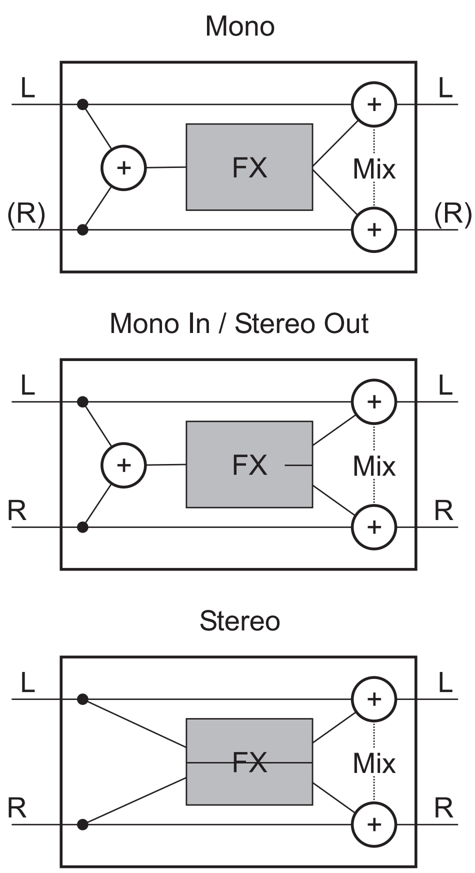
All Parameters from the 8 Effects Units and for all Effects Type are stored in the desk Snapshots.

BSS® Graphic Equalisers

The 35 BSS® 30-band Graphic Equalizers are permanently assigned to the 32 Output Busses and the three Main Masters. All Parameters from the GEQs 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:



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.

Figure 21-1: FX Processor Configurations.

FX Overview Page

In the Overview Page all eight FX processors are visible at the same time and can also be adjusted. The parameters available for adjustment will depend upon the type of FX which is selected. A description of the Effects and their associated controls is given in the section which starts on page 13 of this chapter.
To enter the FX Overview Page press [MENU] and select the FX Tab.
The vertical white Bars on the Boxes represent the assignment mode : In INSERT mode the white bars are inside (example below LEX 1 = Channel Insert, LEX 2 = Master Insert). In PATCH mode the white bars are outside (exampled below LEX 3 = Patch). Note that the bars should both be outside or should both be inside.
The vertical white bars also indicate for each FX processor if a mono or stereo format is being used. In the example below LEX 1 is in a stereo format, all the others are in a mono format.

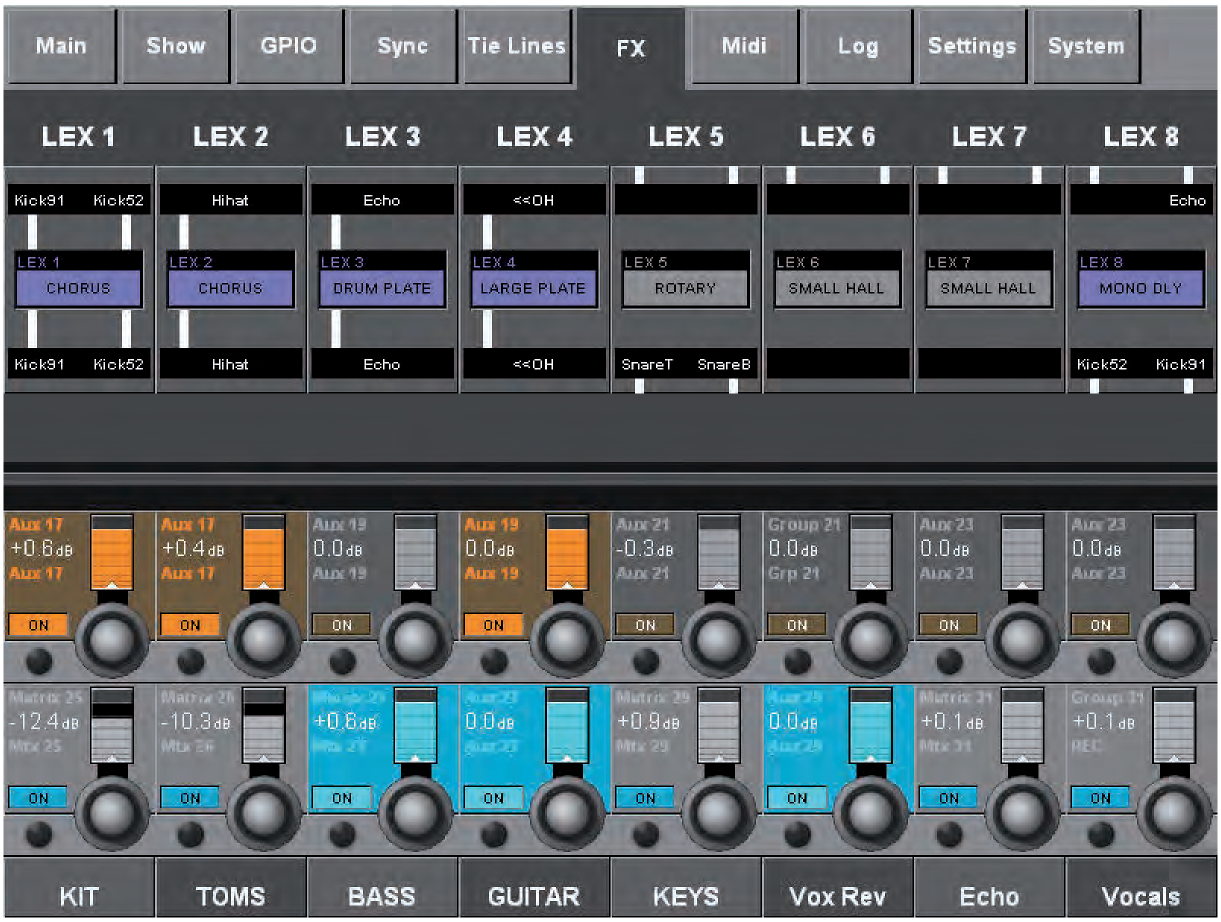


Figure 21-2: FX Overview Page.

HINT. The assignment of the FX processors is visible, but cannot be changed from this page.

It is recommended that before assigning any FX processor this page should be viewed to find out what processors are free (if any). If it is necessary to unassign a processor in order to use it somewhere else it is strongly recommended that the user should unassign all patches to it before re-patching it in its new location.

Snapshot integration

All Parameters from all Effect Types for each of the eight processors are stored in the console Snapshots. In the Basic implementation each of the 8 FX processors can be fully isolated.

TAP

For each effect that offers TAP Tempo (Tempo synchronisation using key press), the bottom left key is used as the TAP button.

ASSIGNING F1-6 KEYS TO FX TAP TEMPO (V2.0 Software and above)

In live situations it is often advantageous to be able to easily control the TAP function from a large button which is permanently accessible on the console surface. The large F1-6 keys below the Master screen can now be used for this purpose.

Figure 21-3: ASSIGNING F1-6 KEYS TO FX TAP TEMPO.

From V2.0 software, Virtual GPI and GPO Pins are available in the Local Rack section of the GPIO Page, in addition to the physical Pins that are used for wiring to external equipment.

The Virtual Pins (VGPI and VGPO) can be used as a way of assigning the F-keys and the F-key LEDs to internal functions in the console.

Currently it is possible to assign the F1-6 keys to remotely control the Lexicon TAP buttons in up to six of the Lexicon FX units.

The Tempo signal from the Lexicon units can be assigned to the F-key LEDs, in order to provide a visual indication of the current tempo.

Changing the assignment of F-keys to TAP function

The first 2 Lexicon units LEX1 and LEX2 have their Delay TAP buttons assigned to F1 and F2 by default because these settings are stored within the read-only factory default Shows (updated with V2.0 software release).

In order to assign the TAP functions of more FX units to the F-keys, proceed as follows:

. 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. Up to 6 of the 8 units can be assigned to F1-6 using this method. The settings will be stored when you save the current Show.

Assigning FX processors

Vi Series supports three different ways to patch an FX processor:

· Insert in an Input Channel
· Insert in a Bus Master
· Patch as an FX return

Channel Insert

This Mode is used for Channel effects.

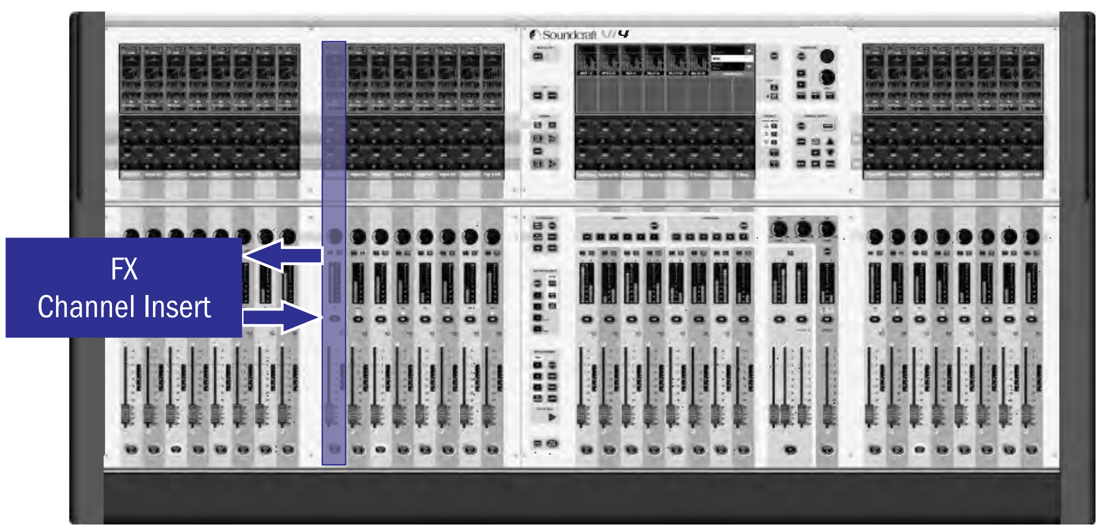


Figure 21-4: FX Channel Insert.

Insert a FX processor in a Input Channel

Touch the VST screen's <PAN> area for the required Input Channel (see Figure 4-13).
Press the {INSERT} key to open the Insert Pool select page (see Figure 21-5).
Pressing the <FX> buton opens the FX selection options.

Figure 21-5: Selecting An FX Processor As An Insert Effect.

Select the desired FX processor. If the processor is in use, a dialog asks if you want to move it from its current location.

Pressing the <EXIT> key will return you to a page similar to Figure 21-6. Notice that the EQ area on the screen is now shared with an FX processor icon.
Pressing the <PAN> area will return the screen to its normal display mode.

The FX processor can be adjusted by pressing the FX processor icon on the screen. This will open a page similar to Figure 21-7. The parameters available for adjustment will depend upon the type of FX which is selected. A description of the Effects and their associated controls is given in the section which starts on page 13 of this chapter.

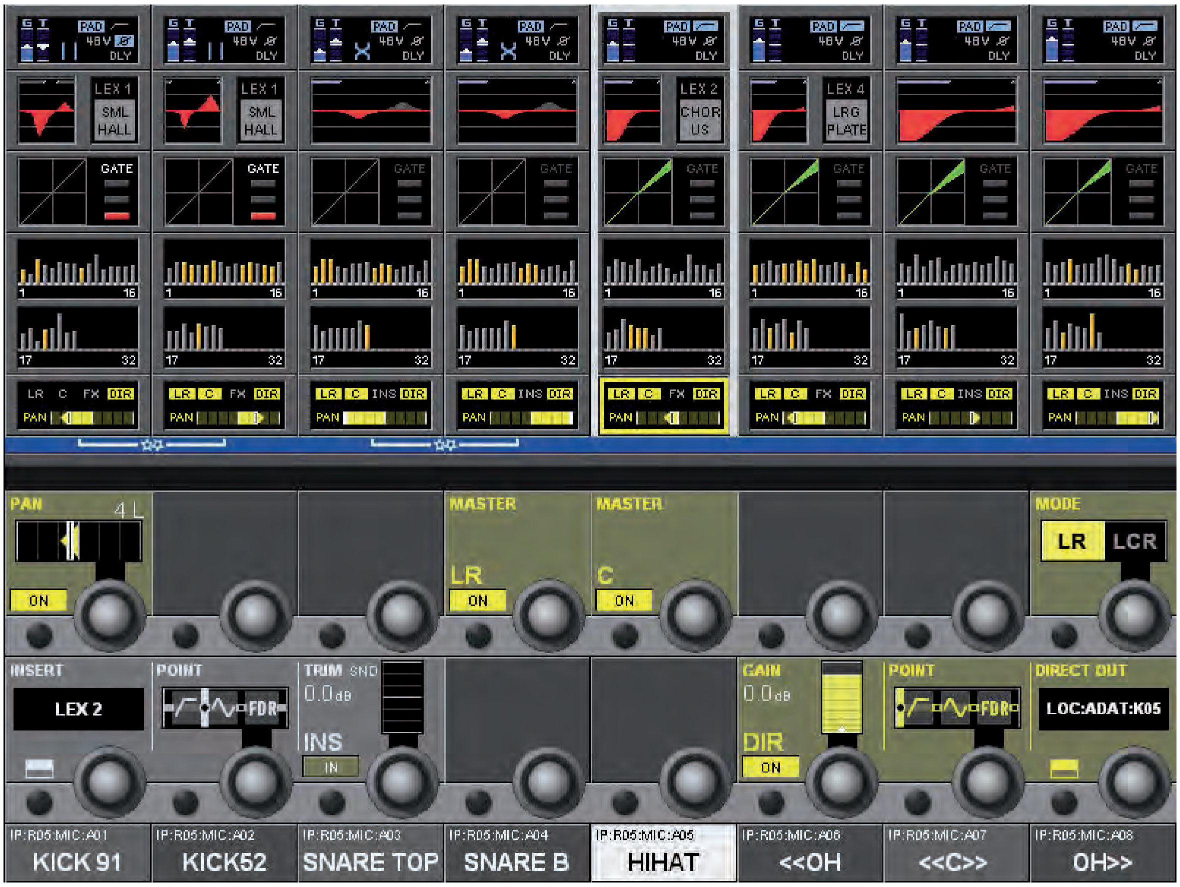


Figure 21-6: An FX Processor As An Insert Effect.

Figure 21-7: Adjusting An FX Processor.

Master Insert

This operation mode is an elegant way to use Reverb Effects without loosing Input Channels for the Return signals.

In this Mode the FX processor is inserted in an AUX Master, and the AUX Master is assigned to the Main Master (LR).

The Input Gain of the FX can be adjusted with the Insert Send TRIM control and the effect amount can be adjusted with the Master Fader

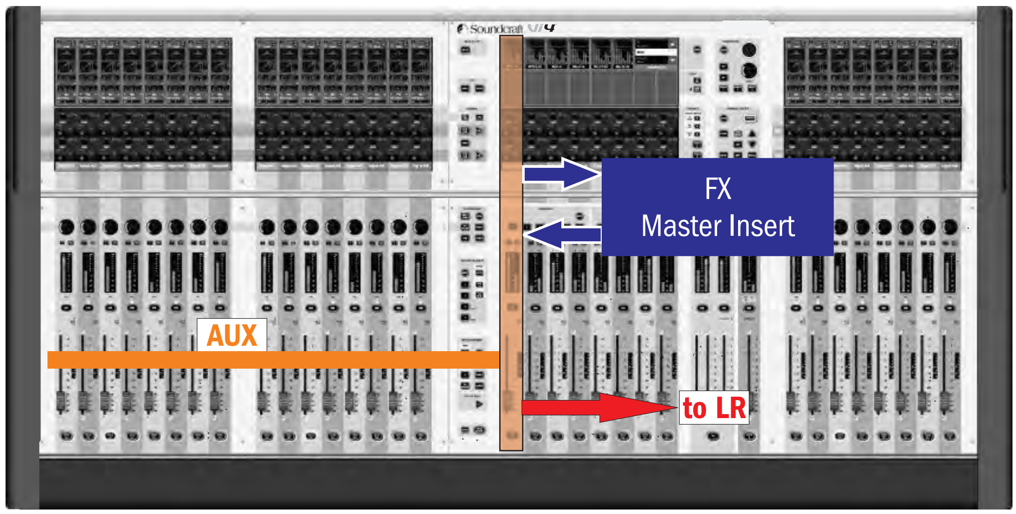


Figure 21-8: FX MASTER INSERT.

HINT

You can use the AUX Master like an FX Return, because it has EQ and Dynamics that can be used on the output from the FX Units. (You must set the Insert Point to pre processing.)

Inserting an FX processor in a Master Bus

Press the [ALL BUSSES] key.
Press the <PAN> area on the required Master Bus (see Figure 21-9).
Press the {INSERT} key to open the Insert Pool select page (see Figure 21-5).

Pressing the <FX> button opens the FX selection options.
Select the desired FX processor. If the processor is in use, a dialog asks if you want to move it from its current location.
Press <Exit> to return to the page similar to Figure 21-9. Notice that an FX processor icon appears in the FX area of the screen when an FX Processor is allocated to the master bus in question.

Press {LR} to route the FX signal to the Main Master bus.
Select the Insert {POINT} to be pre processing in order to use the EQ and Dynamics on the output of the FX.

Pressing the <PAN> area will return the screen to its normal display mode.

The FX processor can be adjusted by pressing the FX processor icon on the screen. This will open a page similar to Figure 21-10. The parameters available for adjustment will depend upon the type of FX which is selected. A description of the Effects and their associated controls is given in the section which starts on page 13 of this chapter.

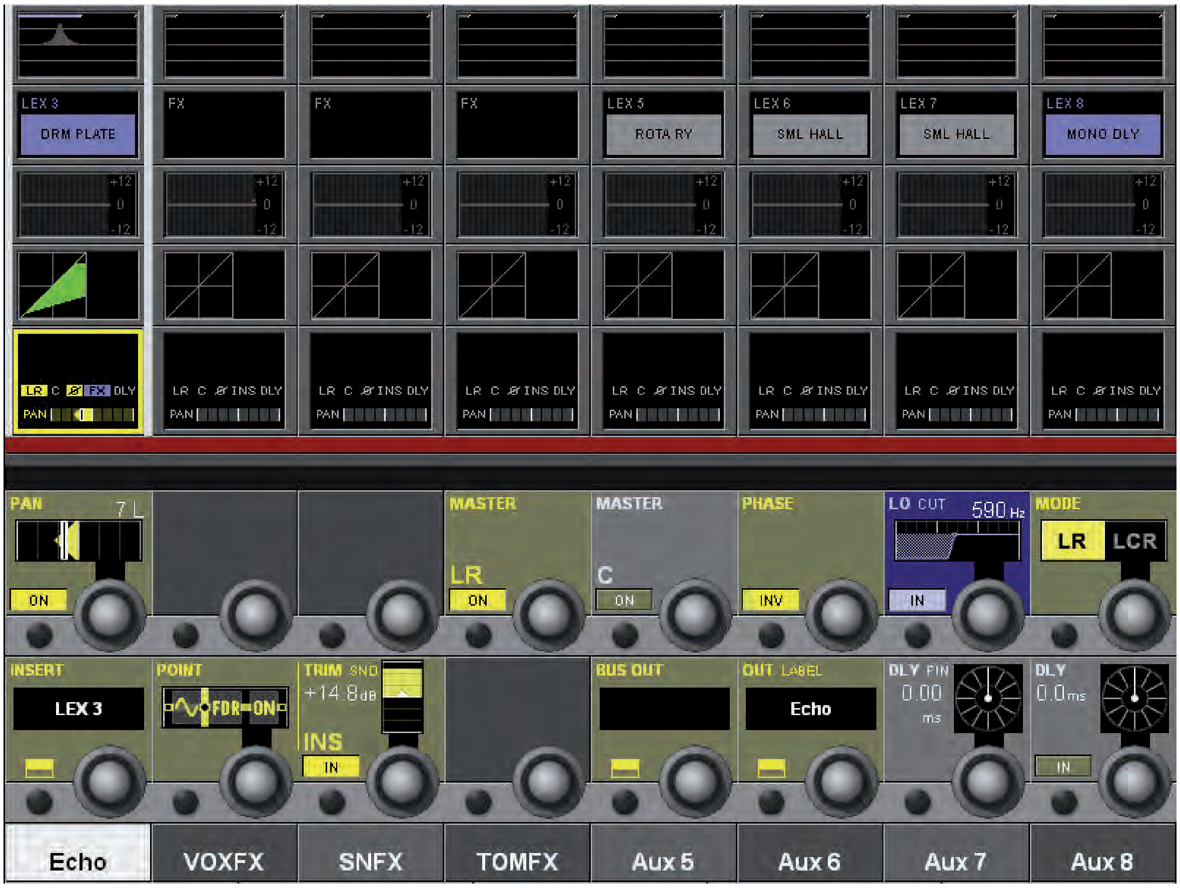


Figure 21-9: Selecting An FX Processor As A Master Bus Insert Effect.

Figure 21-10: Adjusting An FX Processor.

Return in Channel section

This is the classical operation mode for Reverb Effects. The Output of an AUX Master is patched to the Input of an FX Unit, and the Output of the FX Unit is patched to either a mono or 2 paired (Stereo) Input Channels that mixes the Reverb content to the Main Masters or other destinations.

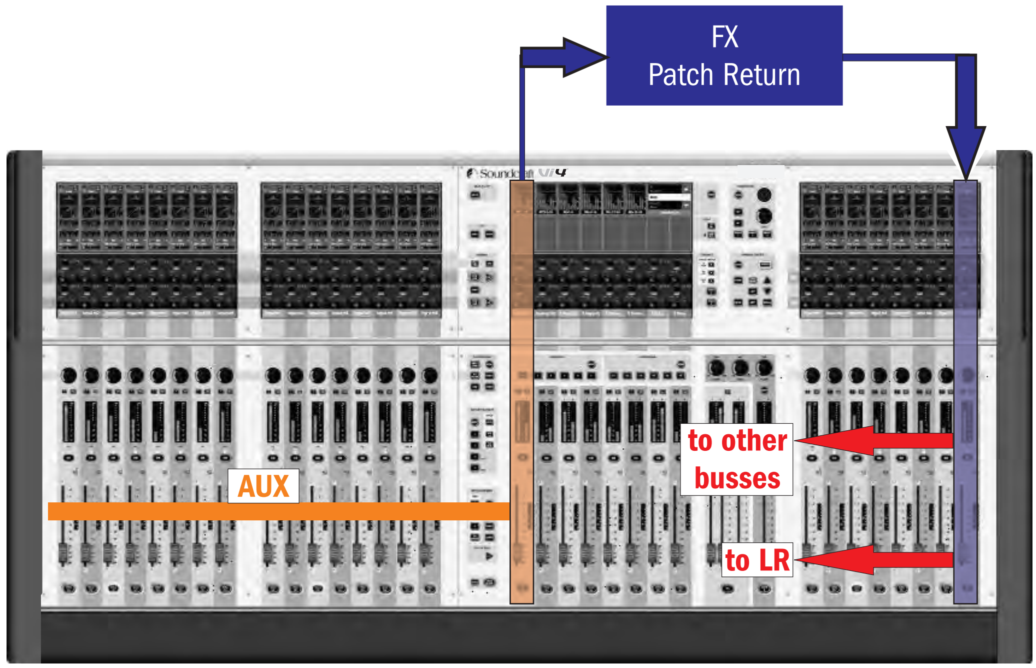


Figure 21-11: FX Return Via An Input Channel.

Patch an FX processor from an AUX Master to the Input section

Press the [ALL BUSSES] key.
Press the <PAN> area on the desired AUX Master bus.
Press the {BUS OUT} key to open the Output Patch page (see Figure 21-12). Press <Lexicon In> to open the FX selection options.
If the Aux bus is mono, you should select both Left and Right Lexicon In patches.
If the Aux bus is stereo, you should patch the Left bus out to the Left Lexicon In, and the Right bus to the Right Lexicon In.
Select the required FX processor. If the processor is in use, a dialog asks if you want to move it from its current location. Press <EXIT>.

Press the Fixed Fader Page [A] or [B] key to select the required bank of input channels.
Press the <INPUT> area on the desired return Input Channel.
Press {IN1 PATCH} to open the Input Patch page (see Figure 21-13). Press <Lexicon Out> to open the FX selection options.
Select the desired FX processor. If the processor is in use, a dialog asks if you want to move it from its current location. Press <EXIT>.

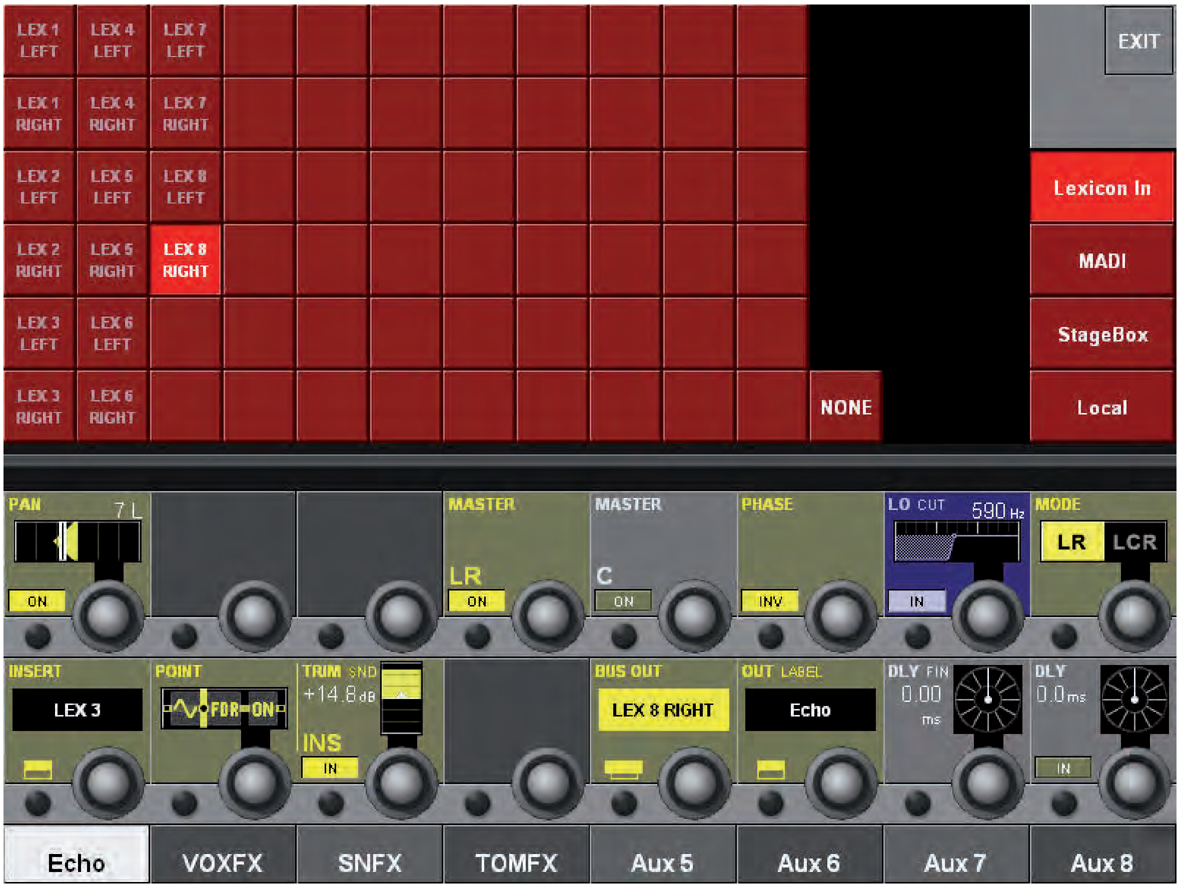


Figure 21-12: Patching An FX Processor To A Master Bus Output.

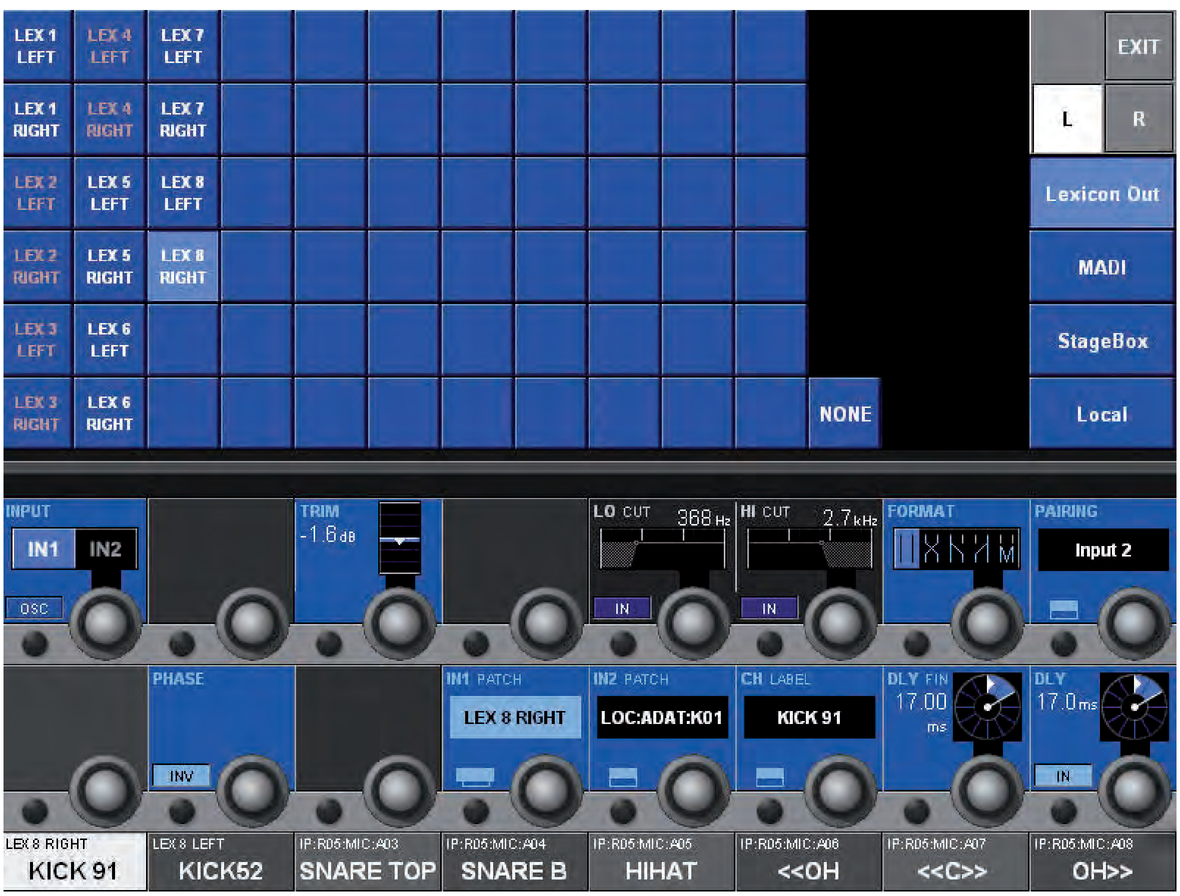


Figure 21-13: Patching An FX Processor To An Input Channel's Input.

FX TYPE

For each of the 8 FX processors an individual FX Type can be selected.
The FX Types are grouped into the following categories:

- · REVERB
- · DELAY
- · MISC

A description of the Effects and their associated controls is given in the section which starts on page 13 of this chapter.

HINT: ***Selecting an FX Type always loads the last user parameter settings for this Type.***

Figure 21-14: Type Selection Page (Reverb).

This page is opened by pressing the {TYPE} button.

Figure 21-15: Type Selection Page (Delay).

Figure 21-16: Type Selection Page (Misc).

FX DESCRIPTIONS

The Vi Series Lexicon FX are divided into three categories: REVERBS, DELAYS and MISC.

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.

Hall Reverbs - *Stereo*
SMALL HALL, LARGE HALL, DRUM HALL, VOCAL HALL.

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 - *Stereo*
SMALL PLATE, LARGE PLATE, DRUM PLATE, VOCAL PLATE.

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 Vi Series FX units model the sound of metal plates with high initial diffusion and a relatively bright, colored 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 color 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 color 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.

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.

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.

Gated Reverb - *Mono In/Stereo Out*
Gated reverb is created by feeding a reverb, such as a metal plate, through a gate device. Decay Time is set to instant, while Hold Time varies duration and sound. The **Gated** reverb provides a fairly constant sound with no decay until the reverb is cut off abruptly. This program works well on percussion - particularly on snare and toms.

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.

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.

Reverb Controls
Pre Delay
Creates an additional 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.

Mid RT
Controls 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
Size sets the 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
Controls the 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 & Spread
In the Hall reverbs, Shape and Spread work together to control the overall ambience of the reverberation. Shape determines 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). 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 buildup and sustain.

RT High Cut
Rt HC sets the frequency above which a 6dB/octave low-pass filter attenuates 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.

Hi Cut
Adjusts 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
Sets 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
Bass Boost boosts or cuts frequencies below Bass Boost Frequency. The amount of boost or cut required is highly dependent on the material being processed.

ER Time
Adjusts the amount of time before reverb early reflections occur.

ER Level
Adjusts the level of early reflections within the reverb.

Feedback Delay
Changing this parameter changes the resonant frequencies of **Plate** reverb.

Feedback Level
Adjusts the **Plate** reverb's presence and prominence.

Boing
This is a unique parameter to the **Spring** reverb, designed to increase or decrease the amount of spring rattle that is a physical characteristic of spring tank reverbs.

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 (center) position.

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 Controls

Tempo

The actual delay time, as tapped in by the **Tempo** button. 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.

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.

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**.

Lo Cut Filter
Frequencies below this level are attenuated.

Hi Cut Filter
Frequencies above this level are attenuated.

Ducker Threshold
The **Studio, Mono** and **Pong** delays offer a "ducking" feature, which causes the delay repeats to be attenuated by a variable amount (between 0 and 18dB) 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
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.

Smear
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.

Level 1 & 2
Adjusts the output level of Tap 1 and Tap 2.

Pan 1 & 2
Adjusts the pan position in the stereo field of Tap 1 and Tap 2.

Mod Depth
This controls 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.

MISC EFFECTS
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 ver y slightly over time. **Chorus** is commonly used to fatten up tracks and to add body to guitars without coloring 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, tunneling, 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/Pan - 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.

Rotary - Mono In/Stereo Out (Wet Only)
Rotary speaker cabinets were designed to provide a majestic vibrato/choir effect for electronic theater 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 modeled 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.

MODULATED EFFECT CONTROLS

Speed

Sets the speed at which the modulated effect cycles.

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.

Voices

Controls the number of additional Chorus voices.

Regen

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.

PreDelay

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.

Waveform

Selects the wave pattern used by the modulated effect.

Phase

Controls whether amplitude or depth change occurs in both left and right outputs simultaneously or alternates between left and right outputs.

Phase Stages

Selects between a 4, 8, or 12 state phase shifter.

Stereo Spread

Increases or decreases the stereo imaging of the Rotary effect.

Drive

Provides overdrive gain to the preamp section of the rotary speaker effect.

Minimum Speed

Sets the minimum speed at which the effect will oscillate.

Maximum Speed
Sets the maximum speed at which the effect will oscillate.

Doppler
Increases or decreases the Doppler pitch effect that is created by the physics of a rotating speaker.

Shift 1 & 2
Determines the amount of pitch shift or detune shift from the original signal source. Works best with individual notes.

Delay 1 & 2
Sets the delay time before the pitch shift or detune effect is heard in the Pitch Shift and Detune effects.

Feedback 1 & 2
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 1 & 2
Sets the pan position in the stereo field for each tap in the 2-Tap Delay.

BSS® Graphic Equalisers

The Vi Series uses a total of 35 high-quality BSS® Graphic Equalisers (GEQ).
Each of the 32 Busses and the three Main Masters are equipped with a 30-band Graphic Equalizer from BSS®. The overall Q of the graphic EQ is adjustable from 4 to 6 in 0.5 steps, to allow a narrower or wider bandwidth of the filers as desired. Narrower bands are better for feedback tuning, while wider bands are more suited to room equalisation.

To access any of the GEQs first press the [ALL BUSSES] key.
A touch on the <GEQ Field> opens the GEQ VST Page and changes the Fader Glow colour of the first 30 Faders to red, this is known as 'LARGE' mode.

It is also possible to have a 'SMALL' control mode where the GEQ is scrolled across only 8 faders to leave input faders available, see page 23.

The first 30 Faders are labelled (small label) in the LCD display with the GEQ Frequencies, and control the Gain of the individual bands. The range of control is +/- 12dB. The cut/boost value of the band is shown in the LCD display and in the Vistonics section when a fader is touched and while being adjusted.

Hint: when a fader is moved away from its default 0dB position, the [ON] key above the fader glows red. Pressing a red [ON] key will reset the fader to the 0dB position.

Figure 21-17 shows an example of the top part of one of the input bay screens.

Figure 21-17: Top Part Of A Graphic EQ Page.

Alternatively, solo an output bus in the Control Bay (ensure that [LOCK MTR] os not ON). This brings up the Output Channel Strip on the Control Bay Vistonics™ screen (see Figure 21-18). Touch the <GEQ> field of the Output Channel to access the GEQ on the faders, as described above.

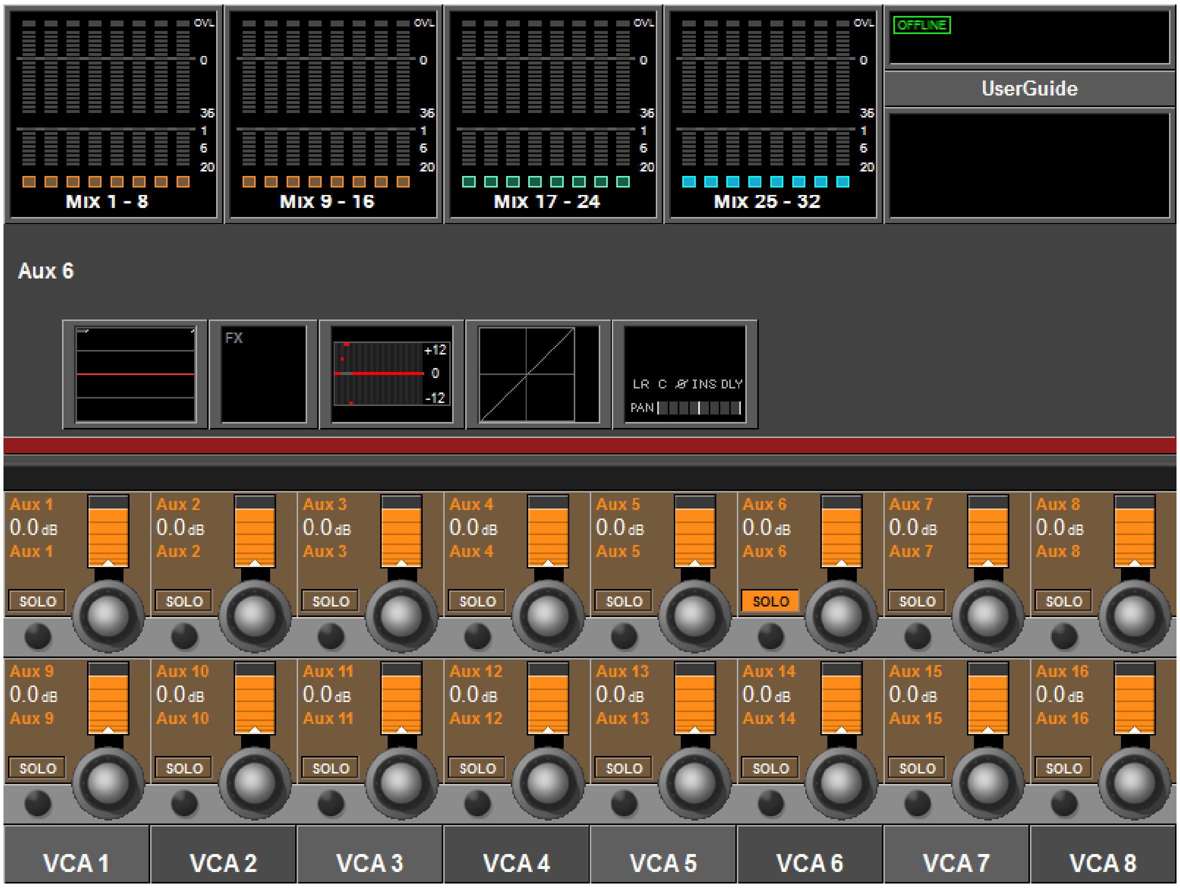


Figure 21-18: Master Bay With An Output Bus Soloed.

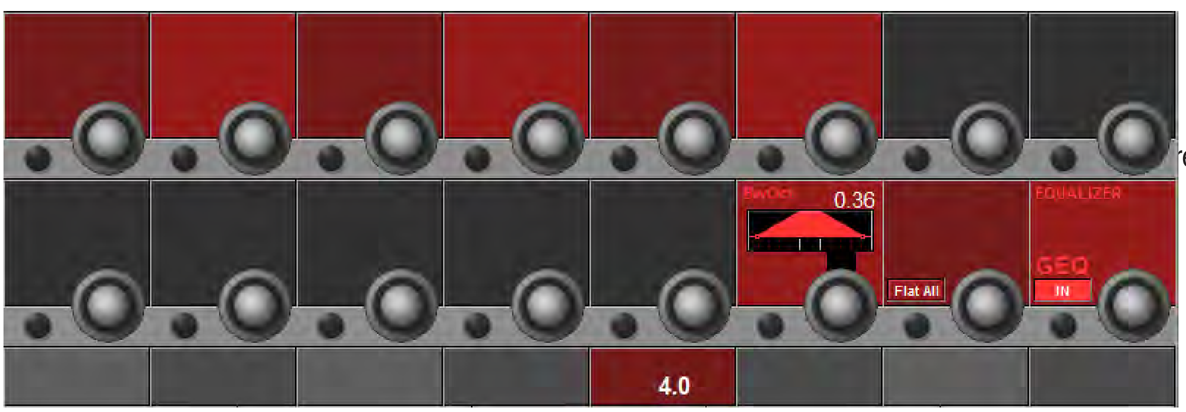
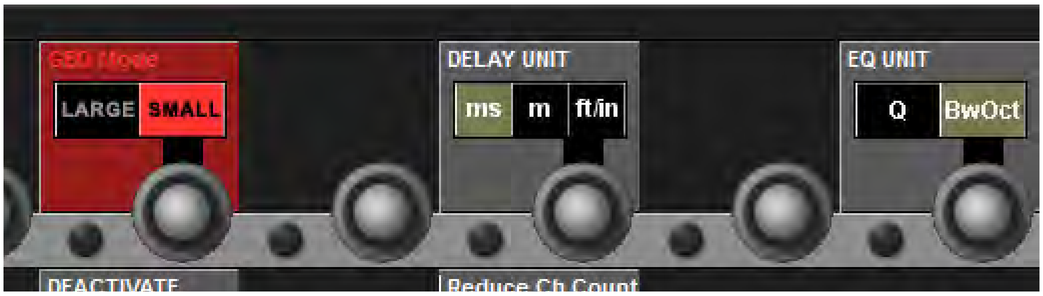


Figure 21-19: Lower Part Of The Master Bay's Screen in Graphic EQ Mode showing cut/boost in Vistonics area when fader is touched.

There are three controls available on this screen:

Width (Q or BWOct)

This allows the operation of the bandwidth controls in the EQ sections throughout the console to be selected as either Octaves or Q-factor. The direction of the control is reversed between the two settings: In Q mode, clockwise narrows bandwidth, in Octaves mode, clockwise widens bandwidth. Settings are saved in the Show file.



The Octaves setting provides a more intuitive control in a musical context

The control adjusts Q between 4 and 6 in 0.5 steps, or bandwidth between 0.24 and 0.36 in 0.03 steps. Selection of Q or BW is made in the SETTINGS Menu.

Note: The setting of the Q/BWOct control affects **ALL** EQ's on the desk, parametric and graphic.

FLAT ALL
{FLAT ALL} set the Gains of all 30 bands to 0 dB

GEQ {IN}
GEQ {IN} switches the GEQ on. The colour of the graph in the <GEQ Field> of the output in question changes to red when it is switched on.

SMALL/LARGE Mode (SETTINGS menu)

Choose 'Large' 30-fader mode for fast access on multiple faders, at the expense of access to the input faders.

Choose 'Small' (8-fader) mode when access to input faders must be retained at all times.

The frequency bands can be scrolled in banks of 4 or 8 bands, using the Output Fader page buttons.

Main Menu – MIDI page open & RX Channel list selected

Figure 22-1: MIDI RX Channels.

The MIDI page of the Main Menu contains the following elements :

- Device Lists for TX MIDI Channel, RX MIDI Channel and TX MIDI Device ID.
- Global MIDI Receive Channel, On/Off and Global Receive MIDI Device ID
- Global MIDI Transmit Channel and On/Off switch.
- MIDI Timecode RX global On/Off switch and Frame Rate control.
- Display Timecode On/Off.

DEVICE LISTS

The Device Lists are mapping tables that display all available MIDI channels (16 for RX, 32 for TX), and allows a user-defined name to be entered for each channel, corresponding to the connected device. There is also a third table that allows 128 Device IDs (similar to MIDI channels but for MMC and MSC messages) to be mapped to names.
x
Since each attached device is allocated its own MIDI channel or Device ID, this allows the user to see the actual device name in the Cue List EDIT MIDI page, making the selection of the required device easier.

By default the Device Name fields are populated with the same text as in the MIDI Channel fields, ie
'Channel 1' – 'Channel 16' (for RX).
'Channel 1 OUT1 – Channel 16 OUT1' and
'Channel 1 OUT2 – Channel 16 OUT2' (for TX)

The settings of the Device names are stored with the Show and are global for the console (not individual per-Cue).

The **<RX CHANNELS>**, **<TX CHANNELS>** and **<TX DEVICE ID>** buttons select one of the three Device Lists for viewing and name editing.

The **NAME** sub-page button opens the QWERTY keyboard and allows the default Device names to be edited.
The **SCROLL Devices** encoder allows fast scrolling of the selected list.

The List wraps around when scrolled – ie it is possible to scroll down from Chan 1 to Chan 16, and up from Chan 16 to Chan 1 in the case of RX, (or from OUT1:1 to OUT2:16 in the case of TX). This is of particular value in the case of the Device ID list, which has a values range from 0 to 127 (127 is displayed as ALL and cannot have its name edited).

Global VST encoders
These encoders are not related to the above Channel Lists. They are located immediately below the lists in order to keep the space on the left clear for future additions.

MIDI IN: The blue area of the screen controls parameters relating to MIDI input.

ON button: works as a global enable buttons for the MIDI IN – when this button is off, no reception is possible from the MIDI IN.

GLOBAL RX CHAN VST encoder: Sets the Global Receive MIDI Channel for the console. This channel can be referenced by the Cue List MIDI RX Channel parameter.
The Global receive channel may also be used in the future for functionality where the console needs to receive MIDI messages that are not associated with the Cue List triggering.
The value set here can nevertheless be selected from within the Cue List MIDI page as one of the available receive channels.
Values range : Off, 1-16, Omni. (Omni mode means receiving on all channels).
Default value: Omni

GLOBAL RX DEVICE ID VST encoder : Sets the Global Receive MID Device ID for the console.
This Device ID can be selected in the Cue List MIDI In page as one of the Device IDs for the MSC 'Goto Cue' trigger messages.
This may also be used in the future to allow a Device ID to be set for the console in case MMC or MSC commands need to be received by the console in future functionality which is not associated with triggering Cues.
Values range : Off, 0-127.
Default value : 127

GLOBAL TIMECODE On/Off VST encoder: Enables or disables the reception of MIDI Timecode. The MIDI Timecode can be set as a trigger for Cues, but if this parameter is set to Off, it will not be possible to recall any Cues with MTC, even if the individual Timecode On/Off switches in each Cue are set to On.

GLOBAL TIMECODE FRAME RATE VST encoder: Allows the frame rate for the internally set values for transmitted and received MIDI Timecode to be set. The frame rate of the incoming timecode timecode for cue triggering is automatically detected.
The frame rate parameter only affects the Frames parameter values range of any Timecode entry fields on the desk (ie: EDIT Cue Number field touched, and EDIT MIDI field touched and a MMC Locate TX event set up).
Values range : discrete values of 24, 25, 30 fps.
Default value : 25

TC Display {ON} key. When switched on, the incoming MIDI timecode value and its automatically detected frame rate is displayed instead of the show name on the main page in the control bay screen.

MIDI OUT: The red area of the screen controls parameters relating to MIDI output.

ON button: works as a global enable buttons for the 2x MIDI OUTs – when this button is off, no transmission is possible from either of the MIDI OUTs.

GLOBAL TX CHAN VST Encoder : Sets the Global Transmit MIDI Channel for the console.
This is not used currently but may be used for future functionality such as the transmission of MIDI messages from the faders and other desk controls.
Values Range : Off, Out1:1-16, Out2:1-16.
Default value: Off.
All parameters in the Menu\MIDI page are stored in the current Show!

Main Menu –MIDI page open & TX Channel list selected

Figure 22-2: MIDI TX Channels.

Showing the Device List with the **Transmit Channels** button selected.

The **NAME** sub-page button opens the QWERTY keyboard and allows the default Device names to be edited.
The unedited names have the format:
OUT1: Channel 1
OUT1: Channel 2
|
V
OUT2: Channel 16

Main Menu –MIDI page open & Transmit Device IDs list selected

Figure 22-3: MIDI TX Device IDs.

Showing the Device List with the **Transmit Device IDs** button selected.

The **NAME** sub-page button opens the QWERTY keyboard and allows the default Device names to be edited.

Device ID 127
With MIDI Device IDs, the number 127 is a special case that is reserved for transmitting to ALL connected devices. Therefore the Device name for the number 127 is fixed as 'ALL' and it is not possible for the User to edit it.

MIDI Event Types

Event Type	Display as:	Value 1, data range Value 2, data range	Cue List	
			TX	RX
Note On	Note On	Note value, 0-127	Yes	Yes
		Velocity 0-127	Yes	No
Program Change	Prog Chng	Program No., 0-127	Yes	Yes
		-		
Controller	Controller	Controller No., 0-127	Yes	Yes
		Value, 0-127		
SysEx String	SysEx	User-defined text string, transmitted as entered.	Yes	No
		-		
MIDI Show Control: GO TO CUE#	Go To Cue	Cue #, No. of cues in the cue list.	Yes	Yes
		-		
MIDI Machine Control (MMC): STOP,PLAY, FF, RW, EJECT, CHASE, PAUSE	MMC Stop MMC Play MMC Pause MMC FF MMC RW MMC Eject MMC Chase	DEVICE ID#, 0-127	Yes	No
		-		
MIDI Machine Control (MMC): LOCATE	MMC Loc	DEVICE ID#, 0-127	Yes	No
		Timecode value, Hrs:min:sec:frames		
MIDI Timecode (MTC)	MTC	Timecode value, Hrs:min:sec:frames	Yes	Yes
		-		

96/72 Channel Version.

On a Vi Series console running V3.0 software or later, the console may be upgraded to 96 (Vi6) or 72 (Vi4) mixing channels, by fitting an additional DSP card to the Local Rack. This upgrade may either be factory fitted at time of order, or can be retrofitted to existing consoles.

This User Guide refers to the standard 64 or 48 channel version in most cases, but the information below describes the differences with the 96/72ch version.

Accessing the Additional 32 (24) channels

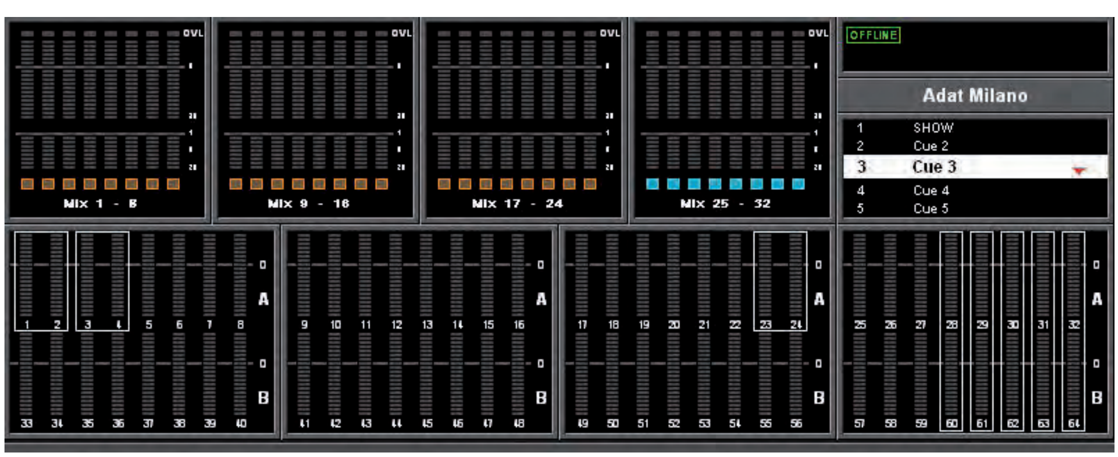
If a third DSP card is fitted to the Local Rack, the C layer button in the Fader page control section of the surface allows access to the extra channels, and the corresponding additional row of input meters input meters will appear in the main screen (see below).

To make full use of the extra channels on the 96ch Vi6, you may wish to add an additional stagebox; this feature was already introduced in the last update (V2.1).

A second stagebox can easily be connected to the spare MADI card in the Local Rack, the only thing to be aware of is that if your desk has a Cat5 Stagebox link, the second MADI card will be optical and therefore you may want to change this card to a Cat5 version, and also purchase an additional breakout panel. Note also that a DIP switch setting will need to be changed in order for the second card to detect a stagebox. See later information for instructions on how to do this.

Busses
The number of busses on the Vi6 remains at 32, with or without the DSP upgrade.

On the Vi4 however, the busses have been increased to 32, which means that if a Vi6 show is loaded onto a Vi4 running V3.0 or later software, all 32 busses will be usable in the same mode as they were on the Vi6 (eg Matrix outs on 25-32).



Hints on using the DSP Upgrade's additional 32/24 channels

When the Vi Series is upgraded with the DSP card addition, there are two ways to use the additional channels:

1. Making more use of the existing I/O

The Vi6 already has 64 mic ins from the stagebox, plus 16 line inputs, 3 mic inputs, 16 AES/EBU inputs and up to 64 MADI inputs in the Local Rack, plus the returns from the 8 built-in stereo Lexicon FX units. This is a total of 179 sources!
This means that there are already plenty of spare physical inputs that can be used to access the additional 32 mixing channels.
Just being able to connect the FX return channels without eating into the Stagebox inputs will be a major benefit.

With this in mind, the new Default Shows provided with the V3.0 and later software have been programmed with the sources for channels 65-80 set to the Local Rack Line inputs.

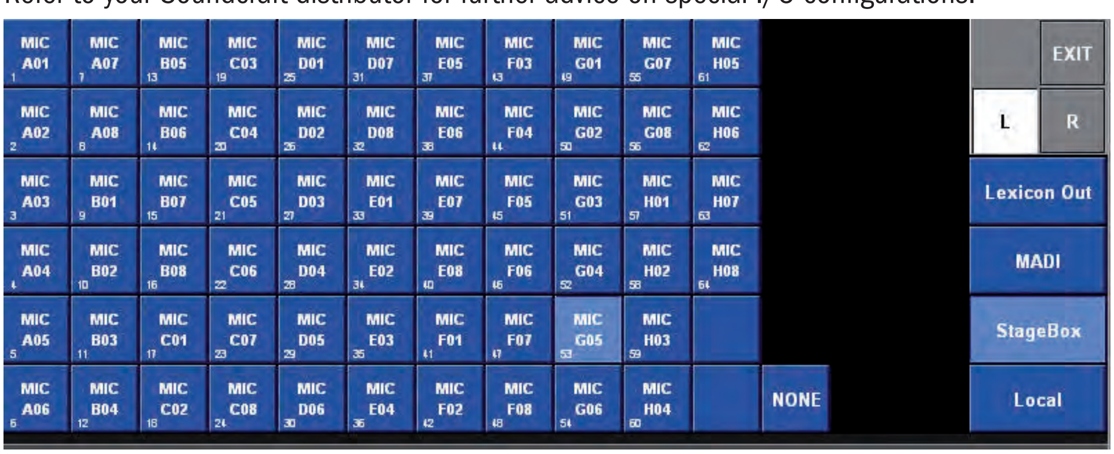
The last 16 channels 81-96 have been left free to use for FX returns since there are different preferences on the vertical or horizontal pairing of these.

2. Add another Stagebox

Alternatively if you require more than 64 inputs from the stage, a second Stagebox can added to the desk, either by connecting it to the existing second MADI card, or adding a third MADI card. (In this case you need to remove the AES/EBU card or 2 line in or line out cards to make room for the MADI card).
As mentioned previously, you may need to obtain additional breakout panels for use with a second Stagebox.

When adding additional MADI cards and Stageboxes, keep in mind that there is a limit to the maximum number of I/O channels that the DSP core can accept – this is 192 inputs and 192 outputs. The total of all the input cards fitted in the local rack must not exceed these numbers. It is possible to add a MADI card and restrict the number of channels it uses, from 64 down to say 32, using internal switches.

Refer to your Soundcraft distributor for further advice on special I/O configurations.



Installing a Third DSP card for 96ch operation

In order to take advantage of the mixing channel upgrade of the V3.0 software, an third DSP card needs to be installed in the Local Rack.

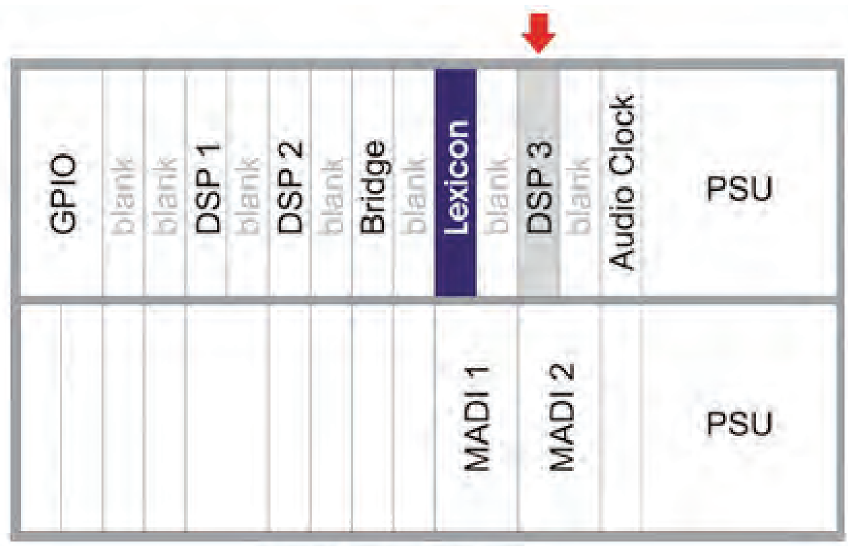
Although DSP cards will work in any of the free slots in the upper section of the Local Rack, it is recommended to fit the additional card in such a way that there is one free slot in between all of the cards, in order to maximize efficient airflow in the rack.

The new DSP card should therefore be fitted to the right of the Lexicon/BSS FX card, as shown in the diagram below (DSP 3 card, indicated with arrow).
Note that the FX card may have to be moved one slot to the left, depending on how it was installed from the factory. If the FX card is already in the position shown below, go straight to step 5.

Instructions for moving FX card and fitting DSP card

Remove any breakout panels to allow access to the top section of the Local Rack.

Note: when removing, refitting and handling DSP and FX cards in the Vi Series rack, observe anti-static precautions: Either wear a grounded wristband or regularly touch the metal chassis of the rack to discharge any static build-up. Keep the new card in its antistatic bag until the last moment.



Local Rack – Rear View

1. Firstly, remove one blanking panel on each side of the FX card.
2. To remove the FX card, unscrew the two small fixing screws at the extreme top and bottom ends of the card (Do not unscrew the two larger extended screws).
3. Release the FX card from the backplane connector by pulling on the two large extended screws. Carefully slide the card from the rack on its mounting rails.
4. Refit the FX card one slot to the left of its previous position. It should then be in the position shown in the diagram above.
5. Fit the new DSP card (DSP 3 in above diagram) to the right of the FX card with one empty space in between.
6. When refitting the FX and DSP cards, engage the pcb carefully with the mounting rails and slide the card into the rack. There should be very little friction if the card is correctly running on the mounting rails. When resistance is felt, press on the two large extended screws to engage the card the last 5mm into the backplane connector. When fully seated, press the floating fascia panel into the rack and tighten the two small fixing screws at top and bottom.
7. The blanking plates should not be refitted until after the FX and DSP cards are fitted, to avoid catching components on the cards on the metalwork as they are slid in and out.

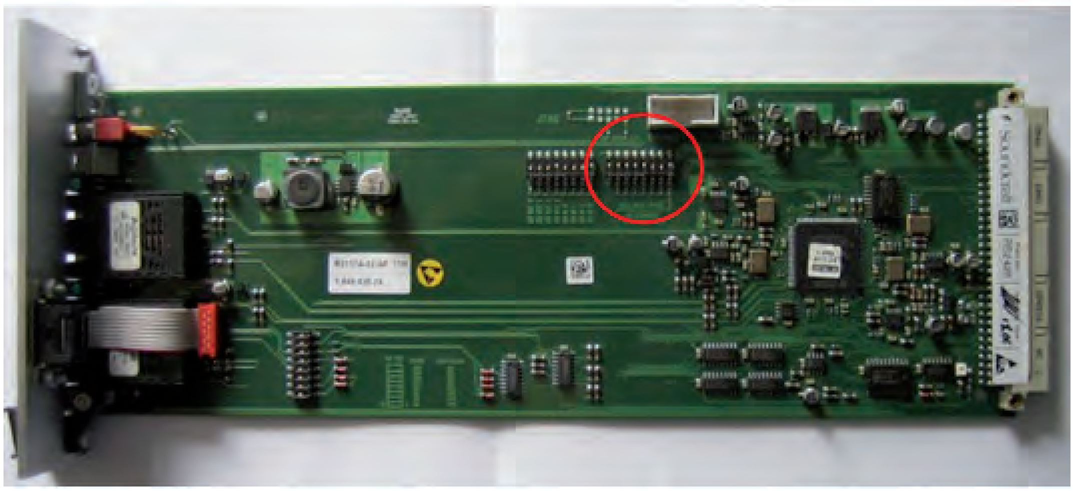
MADI Card Settings for additional Stageboxes

The MADI cards on the Vi4 and 6 contain a DIP switch setting which has to be set differently depending on whether the card is to be used for a Stagebox, or as a simple MADI connection for recording. This setting is necessary because the messages that the card sends out in order to communicate with a stagebox need to be disabled if the card is used to feed a device such as a hard disk recorder, otherwise this can cause problems.

When the consoles are shipped, the first MADI card (left-hand position) is set for Stagebox detection, and the second MADI card (right hand position) is set for Hard Disk recording.
If you want to add another Stagebox and connect this to the second MADI card, the DIP switches on the second card will have to be changed to enable Stagebox detection, as described below.

Checking and Resetting the DIP switches

Remove the optical MADI card from the console, observing anti-static handling procedures.
Locate the required DIP switch using the picture below. The switch has text adjacent to it saying '96k AUX USAGE'.



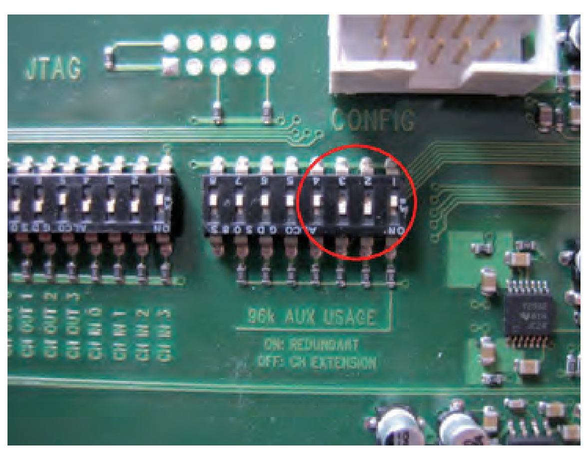
Use a small implement to change the settings of S2 and S3 on this DIP switch as follows:

	Stagebox Detection enabled	Recording Mode (S/Box detection disabled)
S2	OFF	ON
S3	OFF	ON

All other switches should be left in the OFF position.

NOTE: the DIP switch is fitted upside down in relation to the text on the pcb – ensure the correct switch numbers are changed (the numbers and 'ON' position are marked on the switch body, but are very small).

See picture on the right:



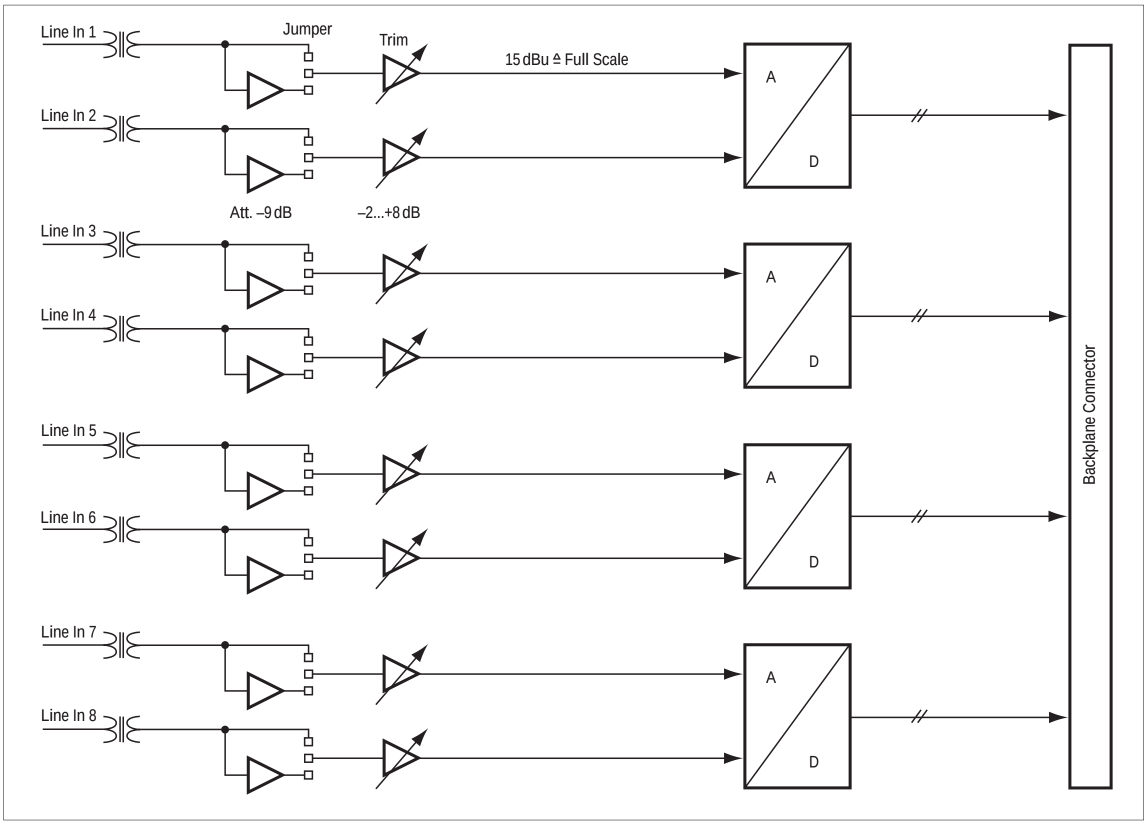
TECHNICAL INFORMATION
Vi Series Standard I/O Cards

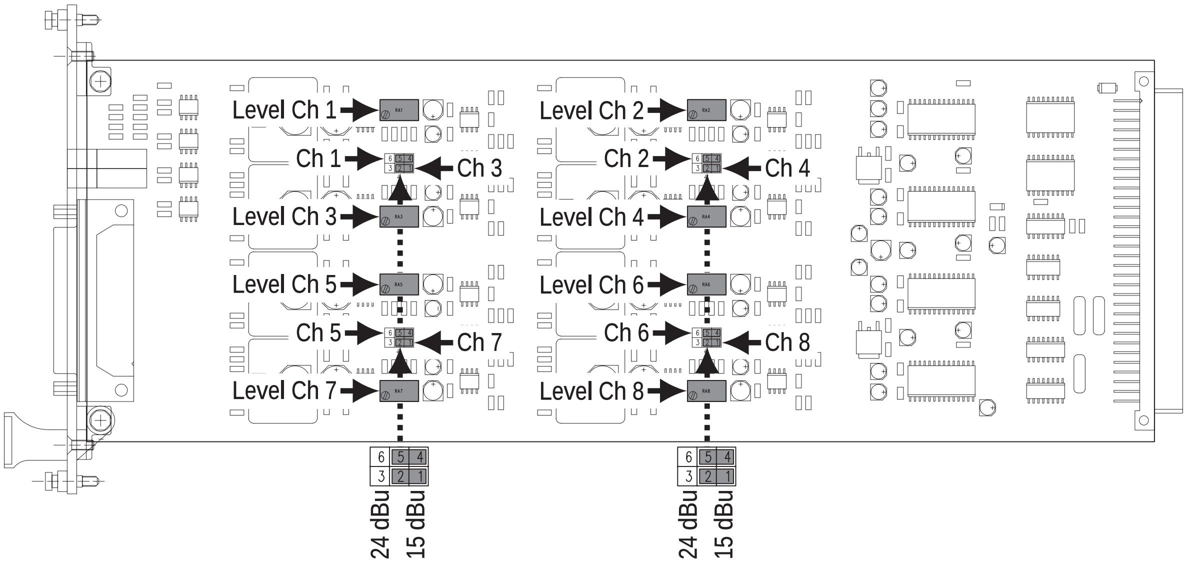
6.2.3 Line In Card

1.949.421

Eight-channel line input card with 24 bit, 44.1/48/88.2/96 kHz A/D Converter, delta-sigma conversion. Transformer-balanced inputs. 96 kHz, 88.2 kHz, 48 kHz, or 44.1 kHz operation. 7...26 dBu input sensitivity. "Signal present" LED indicator. Inputs on standard 25-pin D-type connector (female).

Input level (for 0 dB_{FS})	15/24 dBu (fixed, jumper-selectable), or 7...26 dBu (adjustable)
Input impedance	> 10 kΩ
Frequency response (20 Hz...20 kHz)	–0.2 dB
THD&N (35 Hz...20 kHz, –1 dB_{FS}, min. gain)	< –97 dB_{FS}
(1 kHz, –30 dB_{FS}, min. gain)	< –111 dB_{FS}
Crosstalk (1 kHz)	< –110 dB
Input delay (local)	38 samples (0.79 ms @ 48 kHz)
(remote)	45 samples (0.94 ms @ 48 kHz)
Current consumption (7 V)	0.42 A
(±15 V)	0.1 A
Operating temperature	0...40° C





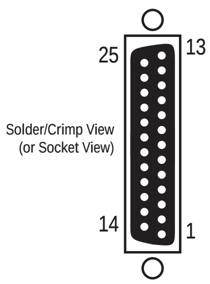
Jumpers: **Level (Ch1...8)** Two positions each: 15 dBu or 24 dBu.

LEDs: **SIGNAL 1...8** For each of the eight channels a green LED indicates if input signal is present; its brightness is a rough indication of the signal level.

Alignment: **RA1...8** *The multi-turn trimmer gives fine adjustment of the input level set with the jumpers. The factory default is +22dBu in =0dBFS.*

If a different input sensitivity has to be adjusted, select the desired range with the jumper and use the LEVEL trimmer potentiometer to adjust to the desired level.
Repeat this alignment for all inputs.

Connector Pin Assignment: (25-pin D-type, female)

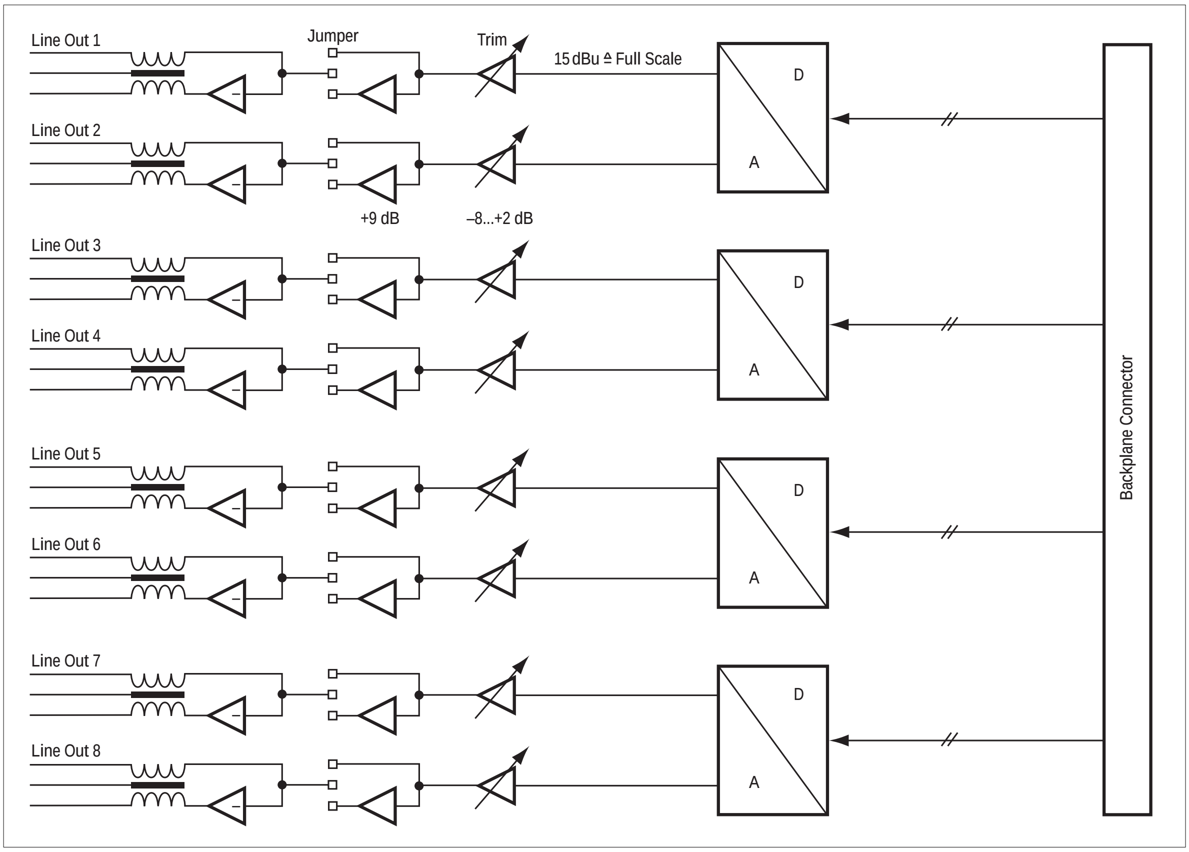


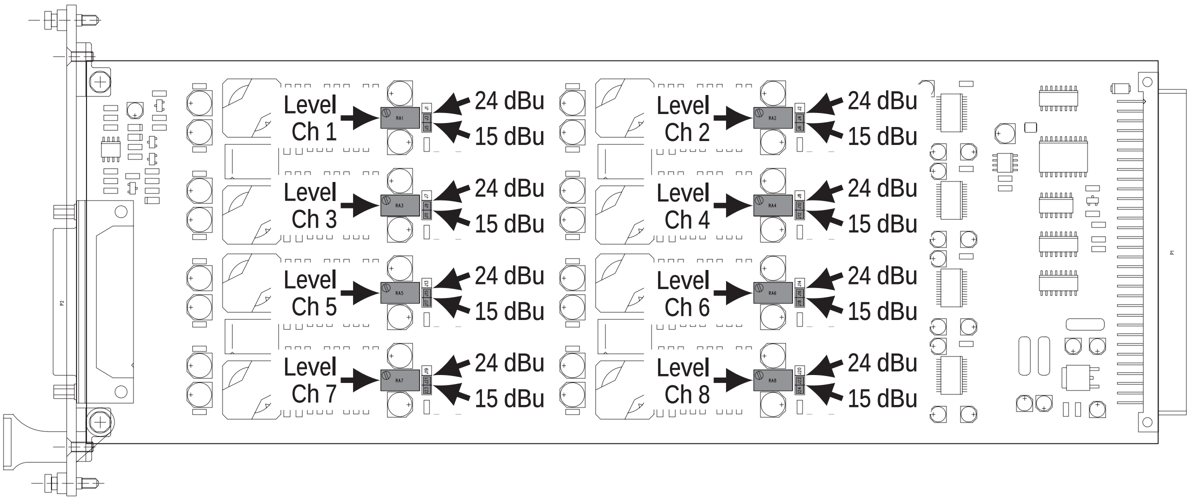
Pin	Signal	Pin	Signal
1	CH 8 in +	14	CH 8 in –
2	CH 8 in GND	15	CH 7 in +
3	CH 7 in –	16	CH 7 in GND
4	CH 6 in +	17	CH 6 in –
5	CH 6 in GND	18	CH 5 in +
6	CH 5 in –	19	CH 5 in GND
7	CH 4 in +	20	CH 4 in –
8	CH 4 in GND	21	CH 3 in +
9	CH 3 in –	22	CH 3 in GND
10	CH 2 in +	23	CH 2 in –
11	CH 2 in GND	24	CH 1 in +
12	CH 1 in –	25	CH 1 in GND
13	n.c.		

6.2.4 Line Out Card 1.949.420

Eight-channel, 24 bit line output card with 24 bit D/A converters with 96 kHz, 88.2 kHz, 48 kHz, or 44.1 kHz operation. Electronically balanced outputs. 7…26 dBu max. output level. Outputs on standard 25-pin D-type connector (female).

Output level (for 0 dB_{FS})	15/24 dBu (fixed, jumper-selectable), or 7...26 dBu (adjustable)
Output impedance	40 Ω
Min. load (at +24 dBu)	600 Ω
Frequency response (20 Hz...20 kHz)	–0.2 dB
THD&N (20 Hz...20 kHz, –1 dB_{FS}, jumper at 15 dBu fixed)	< –90 dB_{FS}
(1 kHz, –30 dB_{FS}, jumper at 15 dBu fixed)	< –110 dB_{FS}
Crosstalk (1 kHz)	< –110 dB
Output delay (local)	28 samples (0.58 ms @ 48 kHz)
(remote)	32 samples (0.67 ms @ 48 kHz)
Current consumption (7 V)	0.23 A
(±15 V)	0.25 A
Operating temperature	0...40° C



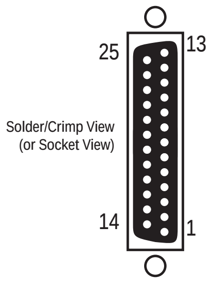


Jumpers: **Level (Ch1...8)** Two positions each: 15 dBu (factory default) or 24 dBu.

Alignment: **RA1...8** *The multi-turn trimmer gives fine adjustment of the output level set with the jumpers. The factory default is +22dBu out = 0dBFS.*

If a different output level is required, select the desired range with the jumper and use the LEVEL trimmer potentiometer to adjust to the desired level.
Repeat this alignment for all outputs.

Connector Pin Assignment: (25-pin D-type, female)



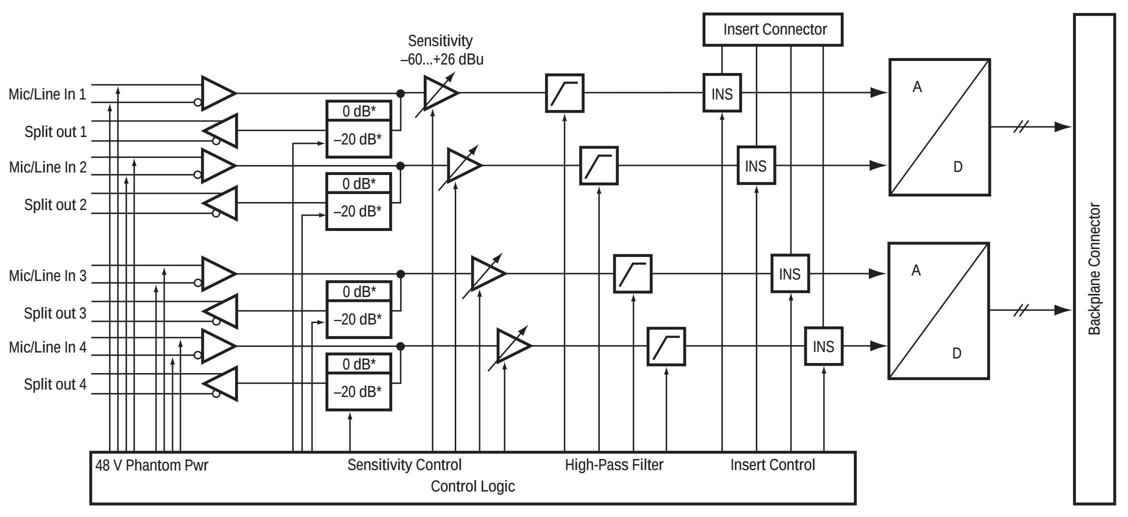
Pin	Signal	Pin	Signal
1	CH 8 out +	14	CH 8 out –
2	CH 8 out GND	15	CH 7 out +
3	CH 7 out –	16	CH 7 out GND
4	CH 6 out +	17	CH 6 out –
5	CH 6 out GND	18	CH 5 out +
6	CH 5 out –	19	CH 5 out GND
7	CH 4 out +	20	CH 4 out –
8	CH 4 out GND	21	CH 3 out +
9	CH 3 out –	22	CH 3 out GND
10	CH 2 out +	23	CH 2 out –
11	CH 2 out GND	24	CH 1 out +
12	CH 1 out –	25	CH 1 out GND
13	n.c.		

6.2 Analog I/O Cards

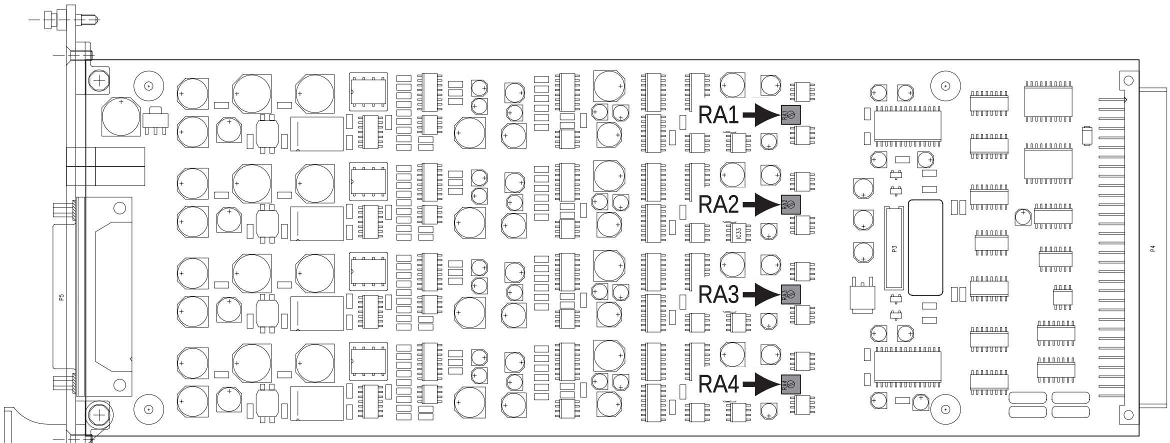
6.2.1 Mic/Line In Card 1.949.427

Four analog microphone/line inputs, electronically balanced, with 24 bit, 44.1/48/88.2/96 kHz delta-sigma A/D converters (mic/line sensitivity, gain setting in 1 dB steps, low-cut filter, soft clipping and 48 V phantom power on/off controlled by console software); four analog split outputs (not used) electronically balanced. Green “signal present” and yellow “phantom power” indicators per channel. Inputs and split outputs on standard 25-pin D-type connector (female).

Input sensitivity (for 0 dB_{FS})	–60…+26 dBu
Input impedance	1.8 kΩ
Equivalent input noise (R_i 200 Ω, max. gain)	–124 dBu
Crosstalk (1 kHz)	< –110 dB
Frequency response (30 Hz...20 kHz)	–0.2 dB
THD&N (35 Hz...20 kHz, –1 dB_{FS}, min. gain)	< –97 dB_{FS}
(1 kHz, –30 dB_{FS}, min. gain)	< –111 dB_{FS}
(input level 6 dBu, min. gain)	< –107 dB_{FS}
CMRR (30 Hz...20 kHz, all gain settings)	> 55 dB
(1 kHz, input sensitivity –10...+26 dBu for 0 dB_{FS})	typ. 100 dB
Low-cut filter	75 Hz / 12 dB/oct.
Input delay (local)	38 samples (0.79 ms @ 48 kHz)
(remote)	45 samples (0.94 ms @ 48 kHz)
Current consumption (7 V)	0.2 A
(±15 V)	0.25 A
Operating temperature	0...40° C



* 0 dB for input sensitivity –60...+3 dBu, –20 dB for input sensitivity +4...+26 dBu

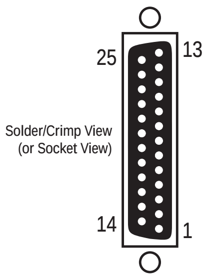


RA1...4: Factory Setting (Level Fine-Adjustment)

LEDs: **PHANTOM 1...4** For each channel a yellow LED indicates if the pantom supply is on.
SIGNAL 1...4 For each channel a green LED indicates if input signal is present; its brightness is a rough indication of the signal level.

Alignment: **RA1...4** *Please note that the input level trimmer potentiometers are factory-set. They need to be adjusted only after having repaired the card.*
Select 15 dBu input sensitivity. Feed an analog signal with a level of +6 dBu to one of the analog inputs. Measure the digital output level either on the MADI output or, after routing through the core, on one of the AES/EBU outputs. Adjust the level with the corresponding FINE ADJUST trimmer potentiometer to –9 dB_{FS}.

Connector Pin Assignment: (25-pin D-type, female)



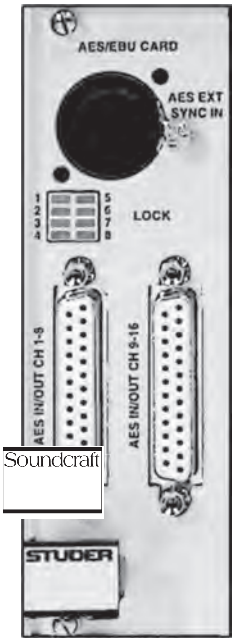
Pin	Signal	Pin	Signal
1	CH 4 split out +	14	CH 4 split out –
2	CH 4 split out GND	15	CH 3 split out +
3	CH 3 split out –	16	CH 3 split out GND
4	CH 2 split out +	17	CH 2 split out –
5	CH 2 split out GND	18	CH 1 split out +
6	CH 1 split out –	19	CH 1 split out GND
7	CH 4 in +	20	CH 4 in –
8	CH 4 in GND	21	CH 3 in +
9	CH 3 in –	22	CH 3 in GND
10	CH 2 in +	23	CH 2 in –
11	CH 2 in GND	24	CH 1 in +
12	CH 1 in –	25	CH 1 in GND
13	n.c.		

Important! If wired correctly, the microphones are isolated from the Local Rack chassis. The circuit inside the microphone takes its supply from pins 2 and 3 (+ and –) for the positive, and from pin 1 (GND) for the negative reference. If a patch bay is implemented, GND (pin 1 on XLR connector) of each microphone input must be connected to its corresponding GND pin, *but not to the chassis*. If chassis instead of GND is used as negative reference for a microphone, it can occur that the GND net of the Local Rack is pulled towards –48 V. This causes the HD link receivers not to work correctly or to be damaged, depending on the type and number of microphones connected.
As a workaround, GND and chassis may be connected inside the Local Rack. frame. In cases where currents flow between the chassis nets of multiple devices, the analog signals can degrade in quality (e.g. perceivable as hum).

6.3 Digital I/O Cards

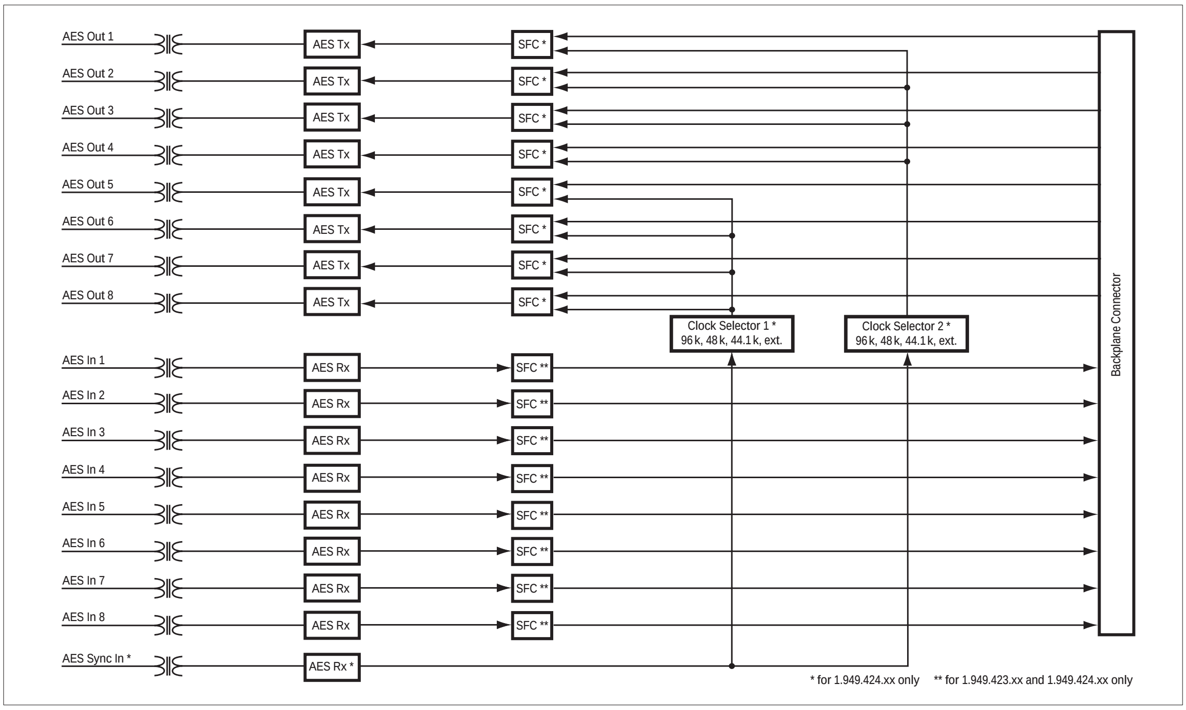
6.3.1 AES/EBU I/O Cards

1.949.422, 1.949.423, 1.949.424

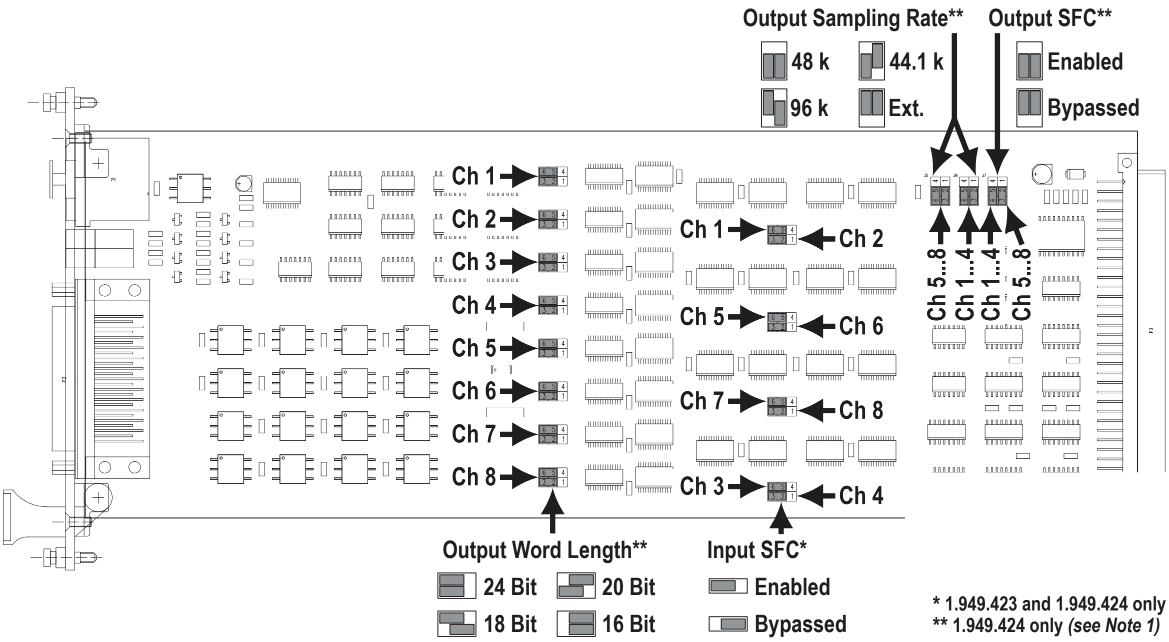


AES/EBU input/output card with 16 Ch I/O.
With input SFCs only.
Selectable output sampling frequencies:
96 kHz, 48 kHz, 44.1 kHz, or external reference (22...108 kHz). Input SFCs can be bypassed individually.

Note: Output SFCs are not fitted on the Vi4/6.

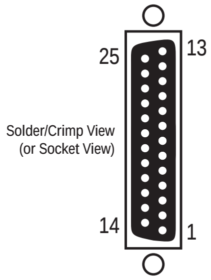


Input / output impedance		110 Ω
Input sensitivity		min. 0.2 V
Output level (into 110 Ω)		5 V
SFC range		22...108 kHz
Current consumption	(3.3 V)	1.949.422: 0.2 A; ..423: 0.4 A; ..424: 0.6 A
	(5 V)	0.65 A
Operating temperature		0...40° C



LEDs: **LOCK 1...8** These green LEDs are on if a valid AES/EBU signal is available at the inputs.

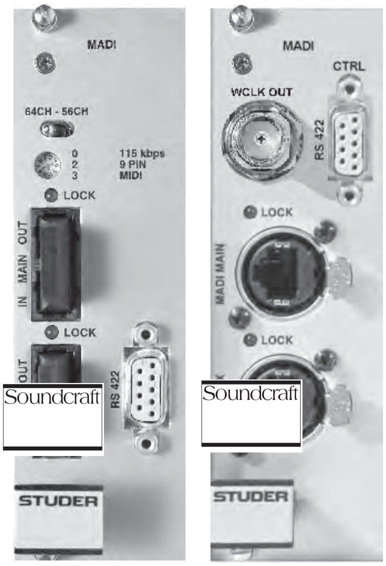
Connector Pin Assignment: (2 × 25-pin D-type, female)



Pin	Signal “CH 1...8”	Signal “CH 9...16”	Pin	Signal “CH 1...8”	Signal “CH 9...16”
1	CH 7/8 out +	CH 15/16 out +	14	CH 7/8 out –	CH 15/16 out –
2	CH 7/8 out screen	CH 15/16 out screen	15	CH 5/6 out +	CH 13/14 out +
3	CH 5/6 out –	CH 13/14 out –	16	CH 5/6 out screen	CH 13/14 out screen
4	CH 3/4 out +	CH 11/12 out +	17	CH 3/4 out –	CH 11/12 out –
5	CH 3/4 out screen	CH 11/12 out screen	18	CH 1/2 out +	CH 9/10 out +
6	CH 1/2 out –	CH 9/10 out –	19	CH 1/2 out screen	CH 9/10 out screen
7	CH 7/8 in +	CH 15/16 in +	20	CH 7/8 in –	CH 15/16 in –
8	CH 7/8 in screen	CH 15/16 in screen	21	CH 5/6 in +	CH 13/14 in +
9	CH 5/6 in –	CH 13/14 in –	22	CH 5/6 in screen	CH 13/14 in screen
10	CH 3/4 in +	CH 11/12 in +	23	CH 3/4 in –	CH 11/12 in –
11	CH 3/4 in screen	CH 11/12 in screen	24	CH 1/2 in +	CH 9/10 in +
12	CH 1/2 in –	CH 9/10 in –	25	CH 1/2 in screen	CH 9/10 in screen
13	n.c.	n.c.			

6.3.2 MADI I/O Cards

1.949.430, 1.949.431, 1.949.433

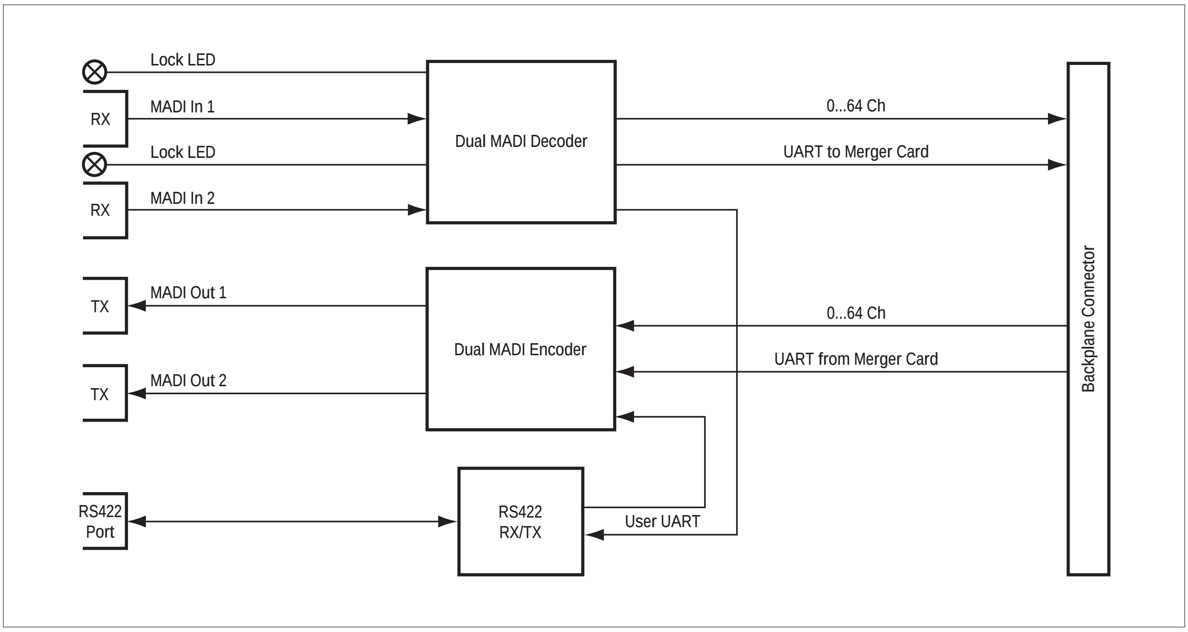


The MADI I/O card can establish a 64-channel MADI input and output to the Stagebox or other MADI-equipped device. Optical inputs and outputs are provided on SC connectors available in multi-mode and single-mode versions, as well as a version with RJ45 connectors for twisted-pair cable and an additional word clock output on a BNC socket.
The auxiliary interface can be used as a redundant link.

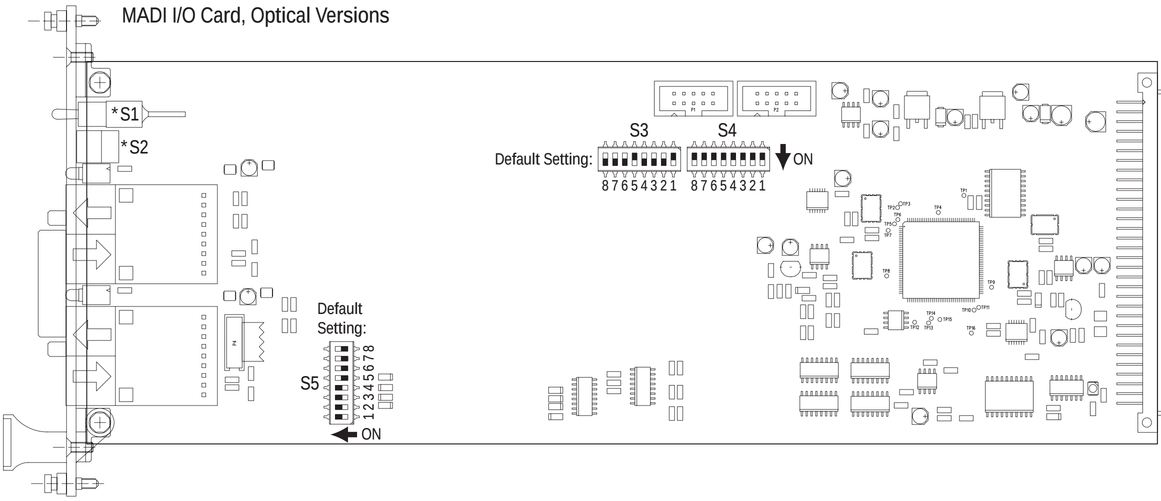
It is possible to transmit any serial control signals, such as MIDI or Sony 9-pin (machine control) through a MADI connection without losing any audio bandwidth or microphone control of the remote I/O box. For this purpose, an RS422 connector is located on this card (hub frame side). The desired baud rate can be set with a rotary switch. The pinout of the RS422 connector can be set to “device” or “controller” with a DIP switch, depending on the 3rd party serial device connected.

Max. cable length	multi-mode fibre, 1300 nm	2 km
	single-mode fibre, 1300 nm	15 km
	CAT5e or better, flexible braid	<75 m
	CAT7, solid core	<120 m

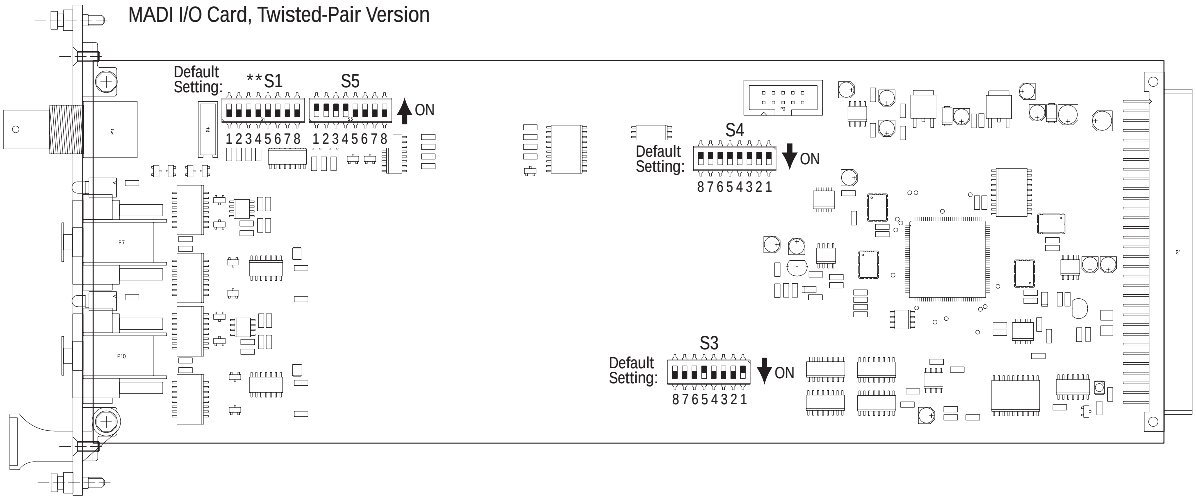
Input frequencies		48kHz ±100 ppm
Current consumption	(3.3 V)	0.4 A
	(5 V)	0.4 A
Operating temperature		0...40° C



MADI I/O Card, Optical Versions



MADI I/O Card, Twisted-Pair Version



Switches: *S1 *(On optical versions only)*

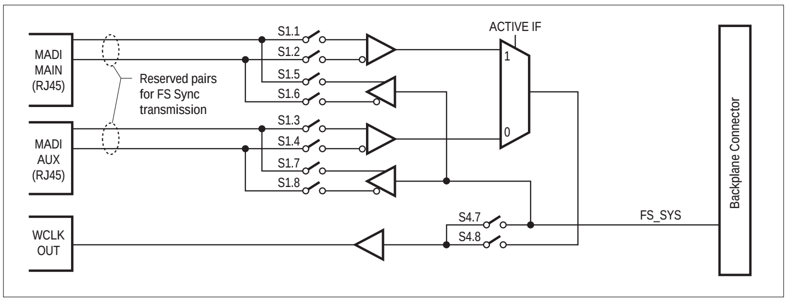
Toggle switch for 64 (factory default) or 56 channel selection.

**S1 *(On Cat5 version only)*

In case of connecting two cores, they must be synchronized. The twisted-pair cable version of the MADI card provides a reserved wire pair for both the main and aux RJ45 sockets on which the sync signal can be transferred. The sync transfer direction (from master to slave) is set using the DIP switches S1 and S4.7/.8. P*lease note that in such a case the twisted-pair wiring has to be done with a crossover cable. On the slave core, the WCLK output must be patched to the WCLK input of the audio clock card.*

(refer to the block diagram on the opposite page)

1	2	3	4	5	6	7	8	Setting
OFF	OFF	OFF	OFF	ON	ON	ON	ON	Card is Master *(factory default)*
ON	ON	ON	ON	OFF	OFF	OFF	OFF	Card is Slave
NO OTHER SETTINGS ALLOWED !								



***S2** *(On optical versions only)*

Rotary switch for baud rate selection of the RS422 user interface:

Position	Setting
0	115'200 bps *(factory default)*
1	57'600 bps
2	38'400 bps (9-pin)
3	31'250 bps (MIDI)
4	19'200 bps
5	9'600 bps
6...9	Reserved for future use

S3 DIP switch for D21m channel count setting:

1	2	3	4	5	6	7	8	Number of Channels
ON	ON	ON	ON	-	-	-	-	0 inputs
ON	ON	ON	OFF	-	-	-	-	8 inputs
ON	ON	OFF	ON	-	-	-	-	16 inputs
ON	ON	OFF	OFF	-	-	-	-	24 inputs
ON	OFF	ON	ON	-	-	-	-	32 inputs
ON	OFF	ON	OFF	-	-	-	-	40 inputs
ON	OFF	OFF	ON	-	-	-	-	48 inputs
ON	OFF	OFF	OFF	-	-	-	-	56 inputs
OFF	ON	ON	ON	-	-	-	-	64 inputs *(factory default)*
OFF : OFF	ON : OFF	ON : OFF	OFF : OFF	- - -	- - -	- - -	- - -	*NOT ALLOWED*
-	-	-	-	ON	ON	ON	ON	0 outputs
-	-	-	-	ON	ON	ON	OFF	8 outputs
-	-	-	-	ON	ON	OFF	ON	16 outputs
-	-	-	-	ON	ON	OFF	OFF	24 outputs
-	-	-	-	ON	OFF	ON	ON	32 outputs
-	-	-	-	ON	OFF	ON	OFF	40 outputs
-	-	-	-	ON	OFF	OFF	ON	48 outputs
-	-	-	-	ON	OFF	OFF	OFF	56 outputs
-	-	-	-	OFF	ON	ON	ON	64 outputs *(factory default)*
- - -	- - -	- - -	- - -	OFF : OFF	ON : OFF	ON : OFF	OFF : OFF	*NOT ALLOWED*

S4 DIP switch for MADI setting *(on the Cat5 version the switches 4...8 are used differently, as indicated below)*:

Card Versions	Switch	Setting
ALL MADI Cards	1	OFF: AUX IF is used for channel extension at 96 kHz *(factory default)* ON: AUX IF is used for redundancy at 96 kHz *(in 48 kHz mode, AUX IF is used for redundancy regardless of the switch setting)*
	2, 3	Both OFF: Standard operation *(factory default)* Both ON: No communication on system UART (used for Hub-Hub interconnection) *One ON and one OFF: NOT ALLOWED.*
Optical Versions only **(RS2426)**	4...7	Must be set to OFF *(factory default)*
	8	Not used *(factory default: OFF)*

Twisted-Pair Cable Version only **(RS2409)**:

4	5	6	Baud Rate
OFF	OFF	OFF	115'200 bps *(factory default)*
OFF	OFF	ON	57'600 bps
OFF	ON	OFF	38'400 bps (9-pin)
OFF	ON	ON	31'250 bps (MIDI)
ON	OFF	OFF	19'200 bps
ON	OFF	ON	9'600 bps
ON ...	ON ...	OFF ...	Reserved for future use

7	8	Setting (refer to **S1 above)
ON	OFF	BNC output carries system word clock *(factory default)*
OFF	ON	BNC output carries received word clock

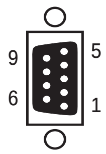
S5 DIP switch for RS422 pinout selection:

1	2	3	4	5	6	7	8	Setting
OFF	OFF	OFF	OFF	ON	ON	ON	ON	RS422 Controller pinout
ON	ON	ON	ON	OFF	OFF	OFF	OFF	RS422 Device pinout *(factory default)*
NO OTHER SETTINGS ALLOWED!								

Connector Pin Assignments:

CTRL (9-pin D-type, female)

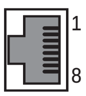
Solder/Crimp View
(or Socket View)



Pin	RS422 Controller	RS422 Device
1	Chassis	Chassis
2	RxD –	TxD –
3	TxD +	RxD +
4	GND	GND
5	n.c.	n.c.
6	GND	GND
7	RxD +	TxD +
8	TxD –	RxD –
9	Chassis	Chassis

MADI MAIN / MADI AUX (8-pin RJ45) *(on Cat5 version only)*

Socket View

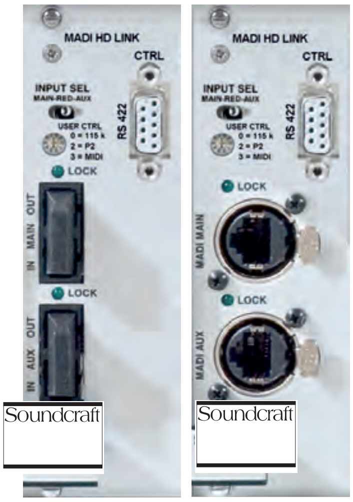


Pin	Signal
1	MADI TxD +
2	MADI TxD –
3	MADI RxD +
4	WCLK TXD/RXD +
5	WCLK TXD/RXD –
6	MADI RxD –
7	reserved
8	reserved

LEDs: On if a valid MADI signal is available at the input that is locked to the system clock.

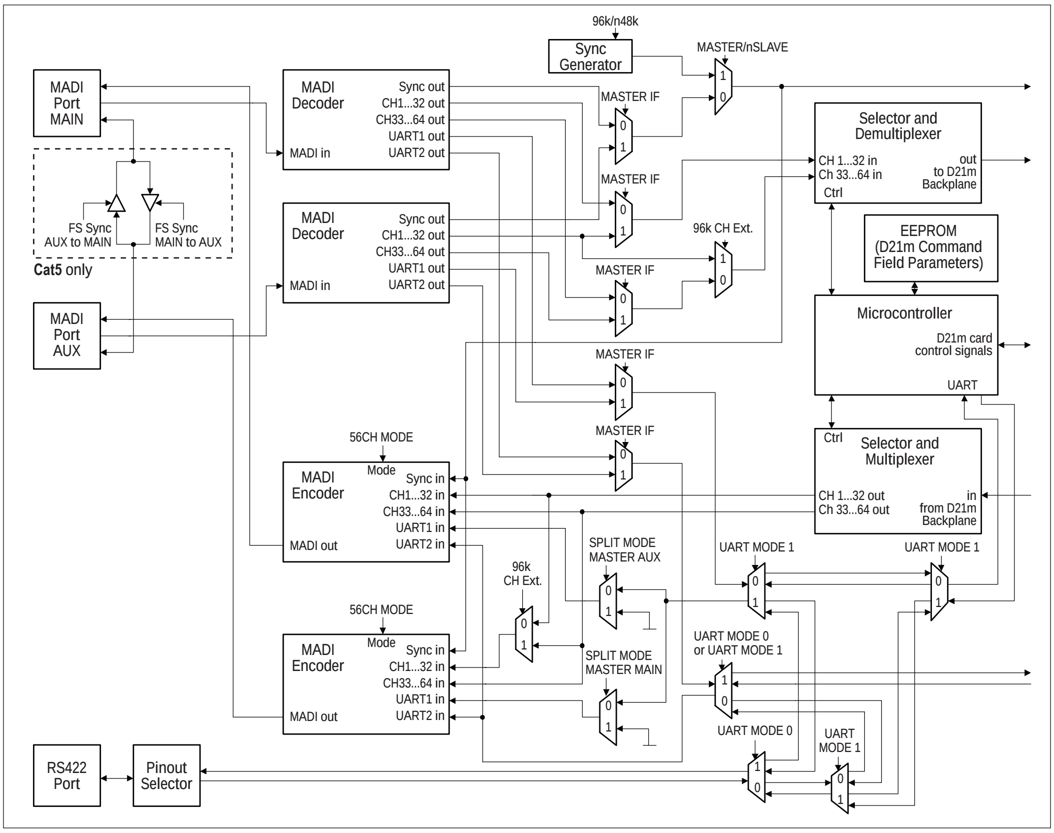
6.5.3 MADI HD Cards

1.949.411, 1.949.413, 1.949.414



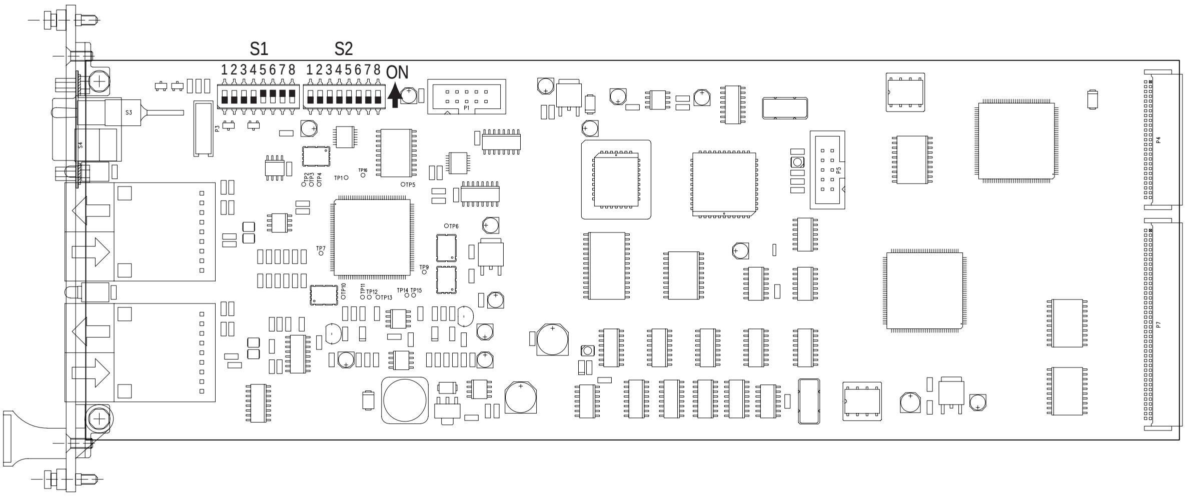
The MADI HD card is plugged into the HD card slot in the Stagebox and provides the link to the Local Rack frame. The two interfaces offer up to 64 audio channels with 48kHz operation, together with embedded control and user-accessible serial connection in each direction. The auxiliary interface can be used as a redundant link.

In slave mode, the card extracts the system clock from the incoming MADI signals and provides it to the entire remote I/O box. It detects all other I/O cards that are inserted into the Stagebox and displays their presence on the front panel of the frame. Once all audio interface cards are plugged in, pressing the **RECONFIG** key on the front panel confirms the configuration to the system. Then all cards are activated and their audio signals are fed into the MADI link.

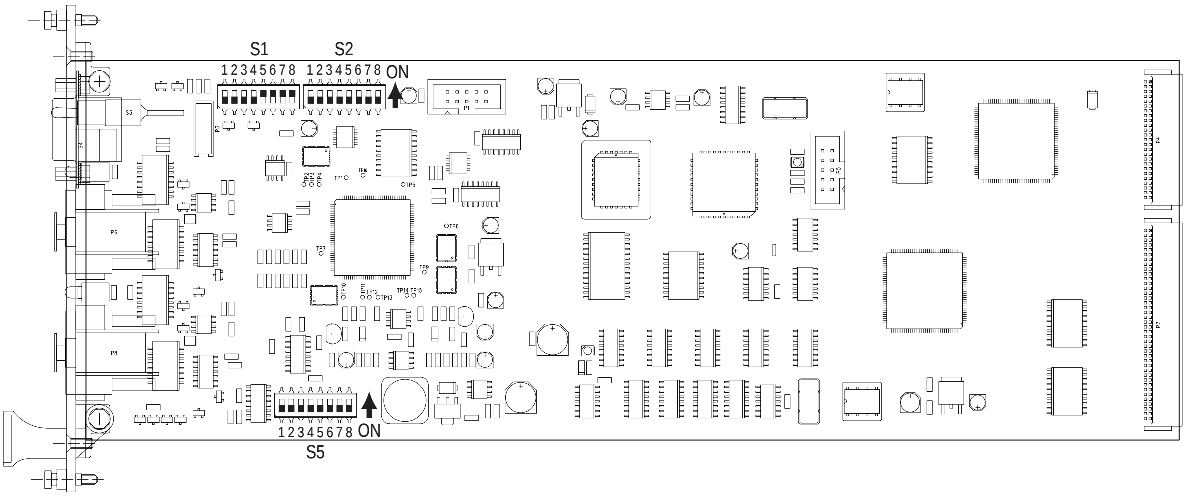


Cable length	multi-mode fibre	<2 km
	single-mode fibre	<15 km (<40 km on request)
	CAT5e or better, flexible braid	<75 m
	CAT7, solid core	<120 m
Input frequency		48kHz ±100 ppm
Current consumption	(3.3 V/5 V)	0.9 A/0.25 A
Operating temperature		0...40° C

Optical



Cat5



LEDs: On if a valid MADI signal is present at the input.

Switches: **S1** DIP switch for pinout selection of the front-panel RS422 connector:

1	2	3	4	5	6	7	8	
ON	ON	ON	ON	OFF	OFF	OFF	OFF	Device pinout
OFF	OFF	OFF	OFF	ON	ON	ON	ON	Controller pinout *(factory default)*
NO OTHER SETTINGS ALLOWED!								

S2 DIP switch for MADI setting:

Switch	Setting		
1	OFF: AUX is used as redundant port at 88.2 / 96 kHz *(factory default)* ON: AUX is used as CH33...64 at 88.2 / 96 kHz		
2	OFF: 64 MADI channels *(factory default)* ON: 56 MADI channels (standard setting for legacy products)		
3, 4	3	4	
	OFF	OFF	MADI1 – Microcontroller / MADI 2 – Front connector *(factory default)*
	ON	OFF	MADI1 – Microcontroller / MADI 2 – Backplane
	OFF	ON	Microcontroller – Front connector / MADI 2 – Backplane
	ON	ON	MADI1 – Front connector / MADI 2 – Backplane
		[Block diagram: UART MODE 1]	
	[Block diagram: UART MODE 0]		
5	OFF: Slave – clock from MADI signal *(factory default)* [Block diagram: MASTER/nSLAVE = 0]		
	ON: Master – clock from local generator [Block diagram: MASTER/nSLAVE = 1]		
6	OFF: Master mode sampling frequency 48 kHz *(factory default)* [Block diagram: 96k/n48k = 0]		
	ON: Master mode sampling frequency 96 kHz [Block diagram: 96k/n48k = 1]		
7, 8	reserved *(factory default: OFF)*		

S3 3-position toggle switch for input selection (MAIN / REDundant / AUX).
MAIN: MADI input is forced to MAIN port (split mode master AUX = 0)
RED: MADI input is used from either MAIN or AUX port
AUX: MADI input is forced to AUX Port (split mode master MAIN = 1).

S4 Rotary switch for baud rate selection of the MADI 2 link:

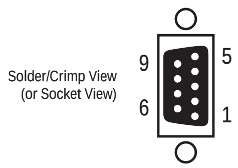
Position	Setting
0	115'200 bps *(factory default)*
1	57'600 bps
2	38'400 bps (9-pin)
3	31'250 bps (MIDI)
4	19'200 bps
5	9'600 bps
6...9	Reserved for future use

S5 DIP switch for FS Sync forward selection *(Cat5 only)*:

1	2	3	4	5	6	7	8	
OFF	OFF	OFF	OFF	OFF	OFF	OFF	OFF	No forward *(factory default)*
ON	ON	ON	ON	OFF	OFF	OFF	OFF	Main to AUX
OFF	OFF	OFF	OFF	ON	ON	ON	ON	AUX to Main
NO OTHER SETTINGS ALLOWED!								

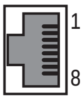
Connector Pin Assignments:

RS422 (9-pin D-type, female)



Pin	RS422 Controller	RS422 Device
1	Chassis	Chassis
2	RxD –	TxD –
3	TxD +	RxD +
4	GND	GND
5	n.c.	n.c.
6	GND	GND
7	RxD +	TxD +
8	TxD –	RxD –
9	Chassis	Chassis

MADI MAIN / MADI AUX (8-pin RJ45)
(on twisted-pair cable version only)



Socket View

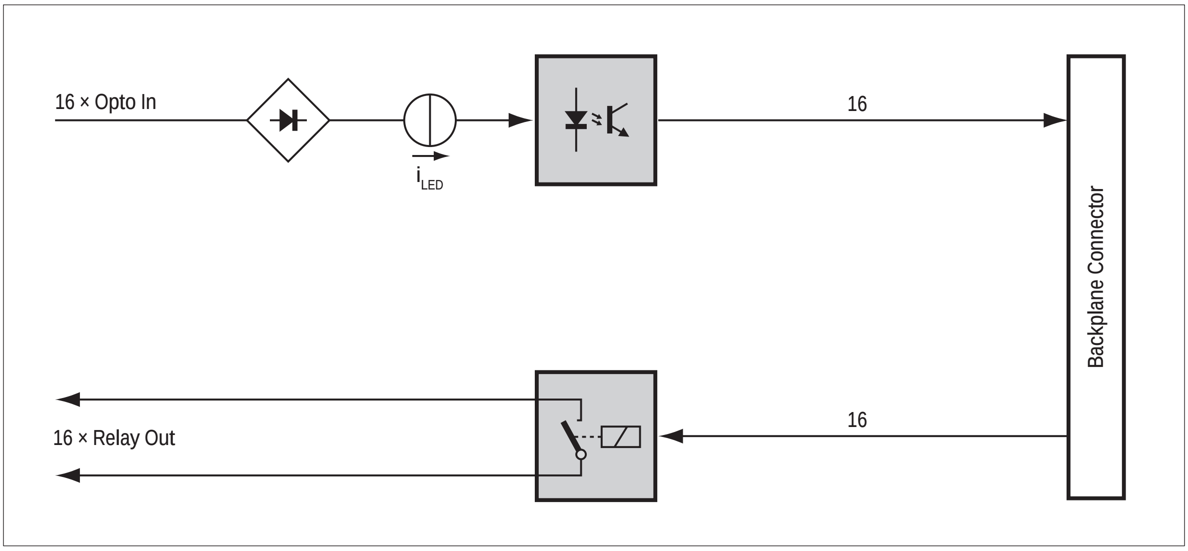
Pin	Signal
1	MADI RxD +
2	MADI RxD –
3	MADI TxD +
4	WCLK TxD/RxD +
5	WCLK TxD/RxD –
6	MADI TxD –
7	reserved
8	reserved

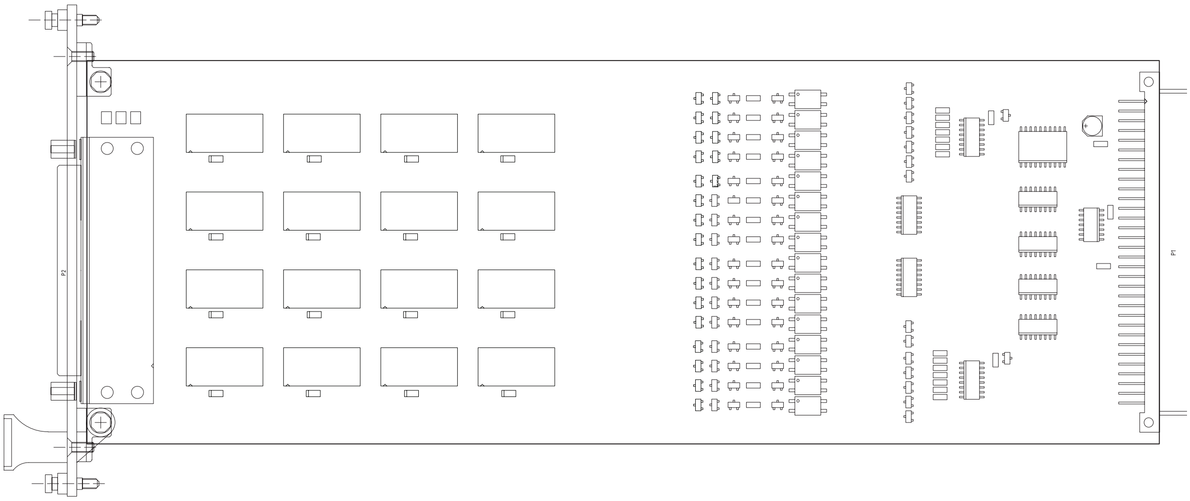
GPIO CARD WITH RELAY OUTPUTS

6.4.2 GPIO Card with Relay Outputs 1.949.436

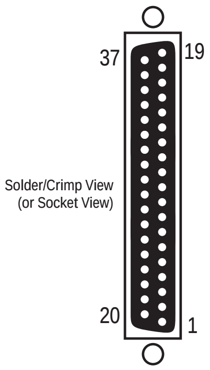
For general-purpose applications requiring total electrical isolation, this card provides 16 electrically isolated opto-coupler inputs with integrated current sink (5...24 V_{DC}) and 16 electrically isolated outputs using SPST relay contacts. 5 V_{DC} supply pins are available. Inputs and outputs on standard 37-pin D-type connectors (female).

Current consumption (5 V)	0.8 A max. (earlier version: 1.1 A max.)
Operating temperature	0...40° C
Output contact rating	0.5 A/125 V_{AC}; 0.7 A/30 V_{DC}; 0.3 A/100 V_{DC}





Connector Pin Assignment: (37-pin D-type, female)



Pin	Signal "GPI 1-16"	Signal "GPO 1-16"	Pin	Signal "GPI 1-16"	Signal "GPO 1-16"
1	GPI 1a	GPO 1a	20	GPI 1b	GPO 1b
2	GPI 2a	GPO 2a	21	GPI 2b	GPO 2b
3	GPI 3a	GPO 3a	22	GPI 3b	GPO 3b
4	GPI 4a	GPO 4a	23	GPI 4b	GPO 4b
5	GPI 5a	GPO 5a	24	GPI 5b	GPO 5b
6	GPI 6a	GPO 6a	25	GPI 6b	GPO 6b
7	GPI 7a	GPO 7a	26	GPI 7b	GPO 7b
8	GPI 8a	GPO 8a	27	GPI 8b	GPO 8b
9	GPI 9a	GPO 9a	28	GPI 9b	GPO 9b
10	GPI 10a	GPO 10a	29	GPI 10b	GPO 10b
11	GPI 11a	GPO 11a	30	GPI 11b	GPO 11b
12	GPI 12a	GPO 12a	31	GPI 12b	GPO 12b
13	GPI 13a	GPO 13a	32	GPI 13b	GPO 13b
14	GPI 14a	GPO 14a	33	GPI 14b	GPO 14b
15	GPI 15a	GPO 15a	34	GPI 15b	GPO 15b
16	GPI 16a	GPO 16a	35	GPI 16b	GPO 16b
17	GND (0 V)	GND (0 V)	36	V_{CC} (+5 V) *	V_{CC} (+5 V) *
18	GND (0 V)	GND (0 V)	37	V_{CC} (+5 V) *	V_{CC} (+5 V) *
19	GND (0 V)	GND (0 V)	** 600 mA max. total*		

Application:

Inputs Control inputs (GPI Xa/b) are completely independent and electrically isolated. They may be used either with the internal +5 V_{DC} supply voltage, or with external voltages of 5...24 V_{DC}, regardless of the polarity. Total current supplied by all +5 V_{DC} pins of one card *must not* exceed 600 mA.

Outputs Control outputs (GPO Xa/b) are completely independent, electrically isolated relay contacts, closed if active. Contact rating is 0.5 A for 125 V_{AC}, 0.7 A for 30 V_{DC}, or 0.3 A for 100 V_{DC}. The +5 V_{DC} supply voltage or the ground (GND) terminals, together with the relay contacts, may be used to generate an output signal. Total current supplied by all +5 V_{DC} pins of one card *must not* exceed 600 mA.

3.4 Ext. Sync Card

1.943.331

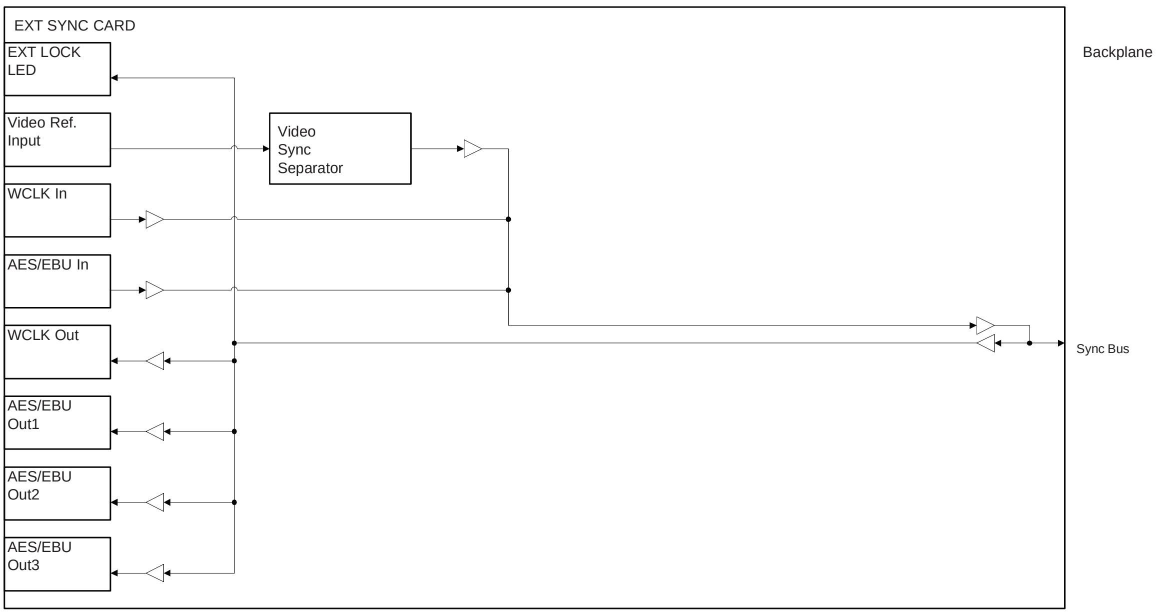
The Ext. Sync card acts as a connector panel for audio clock synchronization inputs and outputs. Three inputs (VIDEO IN, WCLK IN, and AES SYNC IN) can be used; their signals are sent to the sync bus on the backplane. While the video sync is separated on the card, the other signals are just driven onto the bus. A green LED indicates that the applied external sync is used by the system. The system clock can be used from the WCLK OUT and AES 3x OUT outputs.

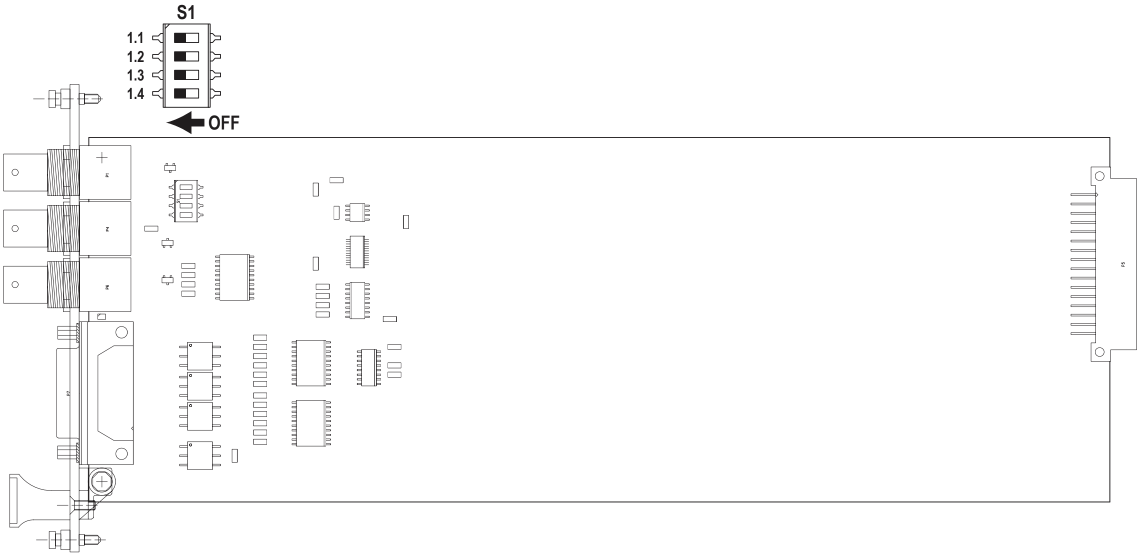
Important: *If the system is using MADI links, the deviation of the external sync signal from the nominal clock frequency **must not exceed ±100 ppm**.* If no MADI links are used, a deviation of ±2500 ppm is tolerated.

Sync Priority The sync source is automatically selected according to the following table:

Clock Source	Priority
Video	1
AES/EBU	2
Wordclock	3
Internal	4

Ext. clock frequency deviation (system *with* MADI links) ±100 ppm
(system *without* MADI links) ±2500 ppm
Current consumption (5 V) approx. 100 mA
Operating temperature 0...40° C



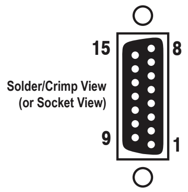


LED: **EXT LOCK** Indicates that the applied external sync is used by the system.

DIP Switch:

S2	Setting
1.1	VIDEO IN termination *(default setting OFF: High-Z / ON: 75 Ω)*
1.2	reserved *(default setting OFF)*
1.3	reserved *(default setting OFF)*
1.4	WCLK IN termination *(default setting OFF: High-Z / ON: 75 Ω)*

Connector Pin Assignment: **AES SYNC IN + 3x OUT** (15-pin D-type, female)



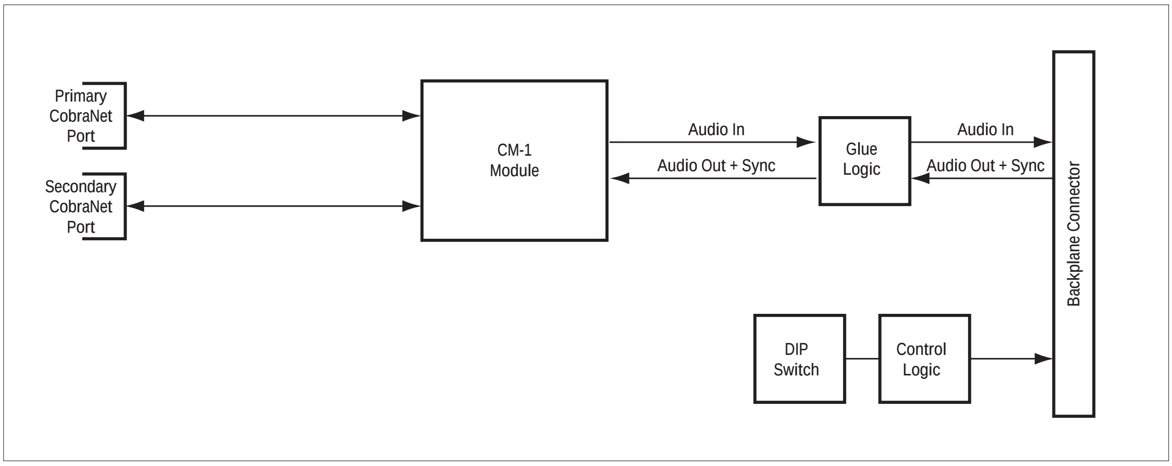
Pin	Signal	Pin	Signal
1	AES IN +	9	AES IN –
2	Screen	10	Screen
3	AES OUT 3 –	11	AES OUT 3 +
4	n.c.	12	n.c.
5	AES OUT 2 +	13	AES OUT 2 –
6	Screen	14	Screen
7	AES OUT 1 –	15	AES OUT 1 +
8	n.c.		

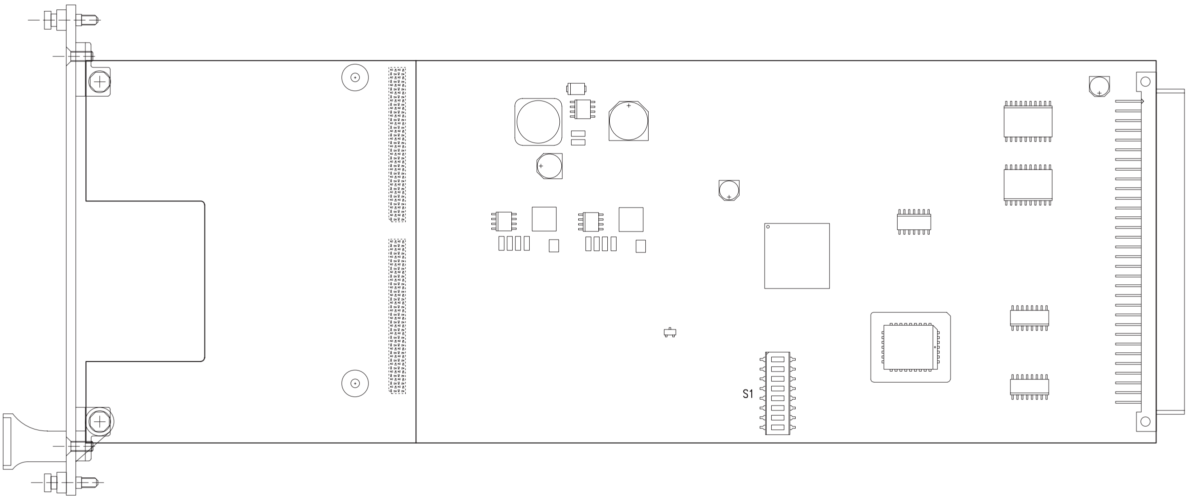
6.3.8 CobraNet® Card

1.949.445

This card allows sending and receiving of up to 32 audio channels to/from a CobraNet®. DIP switches on the card allow setting the number of input or output channels seen by the console. Default setting is 32 output and no input channels. All settings of the CobraNet® module are made through SNMP. By default, the module is configured to be the conductor (synchronization master) and providing unicast bundles 1001...1004 to the CobraNet® network. This setting is ideal for e.g. providing audio channels to a PA, installed sound, or monitoring system using CobraNet®.
For further information on CobraNet®, please refer to the CobraNet® user's manual or to www.cobranet.info.

Current consumption (5 V)	800 mA
Operating temperature	0...40° C





DIP Switch: **S1** DIP switch for channel count setting:

1	2	3	4	5	6	7	8	Number of Channels
OFF	OFF	OFF	OFF	-	-	-	-	0 inputs *(factory default)*
OFF	OFF	OFF	ON	-	-	-	-	8 inputs
OFF	OFF	ON	OFF	-	-	-	-	16 inputs
OFF	OFF	ON	ON	-	-	-	-	24 inputs
OFF	ON	OFF	OFF	-	-	-	-	32 inputs
OFF : ON	ON : ON	OFF : ON	ON : ON	- - -	- - -	- - -	- - -	*NOT ALLOWED*
-	-	-	-	OFF	OFF	OFF	OFF	0 outputs
-	-	-	-	OFF	OFF	OFF	ON	8 outputs
-	-	-	-	OFF	OFF	ON	OFF	16 outputs
-	-	-	-	OFF	OFF	ON	ON	24 outputs
-	-	-	-	OFF	ON	OFF	OFF	32 outputs *(factory default)*
- - -	- - -	- - -	- - -	OFF : ON	ON : ON	OFF : ON	ON : ON	*NOT ALLOWED*

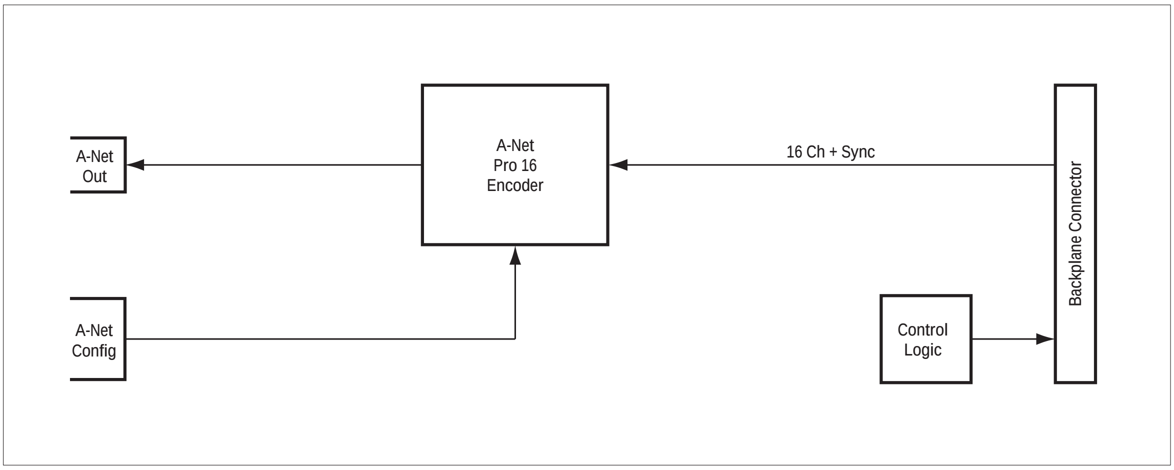
Aviom A-Net® CARD

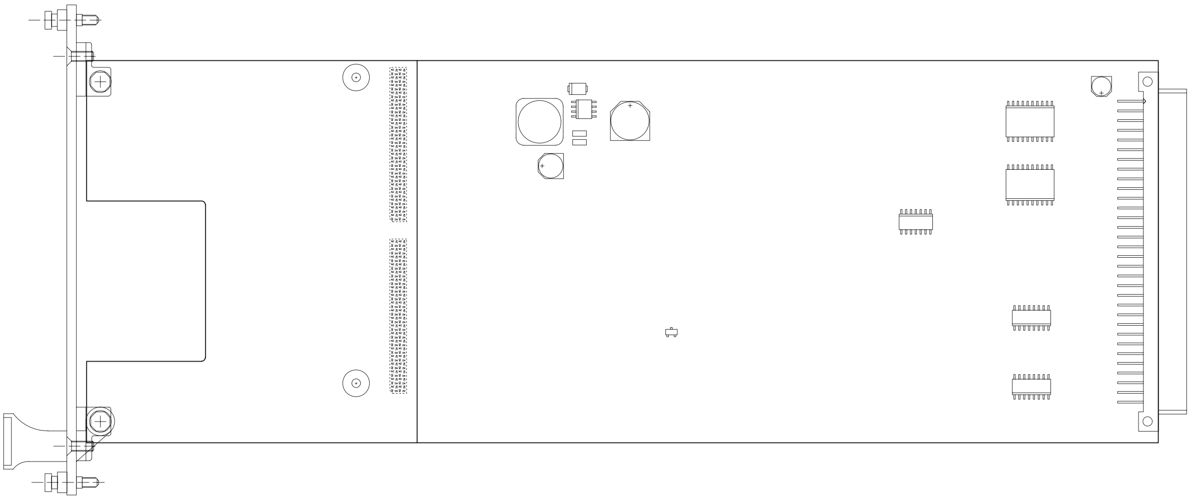
6.3.9 Aviom A-Net® Card

1.949.446

This card allows the Vi desk to digitally feed an Aviom A-Net® Pro-16 chain. With this standard, 16 mono signals can be fed to any number of Aviom personal mixers (such as the A-16 II), connected in a daisy chain configuration. The A-Net® card will be the start of the chain and provide the audio and synchronization data to the chain. DIP switches on the front panel allow grouping two adjacent channels to one stereo channel, and generating a test tone.
The card is available for both Local Rack and Stagebox.

Current consumption (5 V)	250 mA
Operating temperature	0...40° C





Front-Panel Switch:

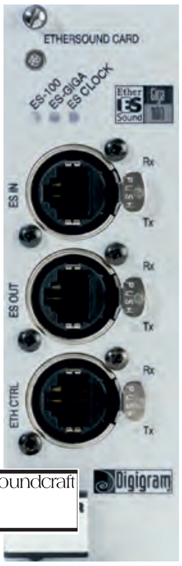
Position	Setting
1	OFF: Channels 1 and 2 are mono *(factory default)* ON: Channels 1 and 2 are a stereo group
2	OFF: Channels 3 and 4 are mono *(factory default)* ON: Channels 3 and 4 are a stereo group
3	OFF: Channels 5 and 6 are mono *(factory default)* ON: Channels 5 and 6 are a stereo group
4	OFF: Channels 7 and 8 are mono *(factory default)* ON: Channels 7 and 8 are a stereo group
5	OFF: Channels 9 and 10 are mono *(factory default)* ON: Channels 9 and 10 are a stereo group
6	OFF: Channels 11 and 12 are mono *(factory default)* ON: Channels 11 and 12 are a stereo group
7	OFF: Channels 13 and 14 are mono *(factory default)* ON: Channels 13 and 14 are a stereo group
8	OFF: Channels 15 and 16 are mono *(factory default)* ON: Channels 15 and 16 are a stereo group
9	OFF: Test tone generator off *(factory default)* ON: Test tone generator on

EtherSound® CARD

Note: only available via Digigram distribution network. Please contact www.digigram.com for further details.

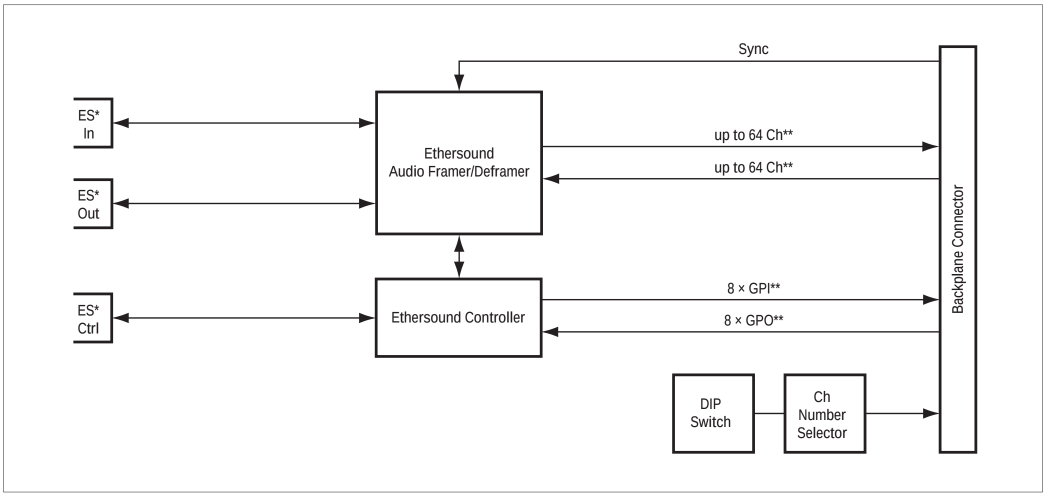
6.3.10 EtherSound® Card

(please contact www.digigram.com for further details)



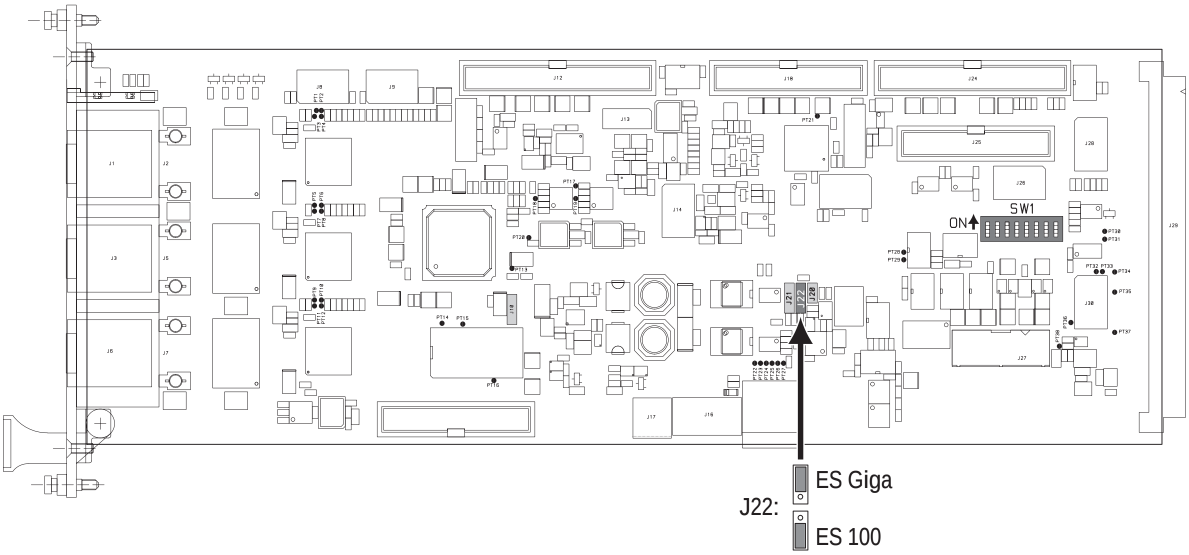
The EtherSound® card allows connecting the Vi console to an Ether-Sound® network. From the desk's viewpoint, it acts in a similar way to a MADI card combined with a GPIO card. The number of audio channels used can be configured with DIP switches. The included, virtual GPIO card allows, e.g., routing a GPO of the mixing console to the GPO of a distant EtherSound® device on the network. Configuration of the EtherSound® network is performed either through the ETH CTRL connector or from a remote location on the EtherSound® network, e.g. using the EtherSound® EScontrol software.
The EtherSound® card works with EtherSound® ES-Giga System Transport networks or with EtherSound® ES-100 Audio Transport networks. The operating mode of the card (ES-100 or ES-Giga) is selected by setting jumper J22 (see opposite page). The selected mode will be displayed on the front panel LEDs.
The audio clock of the EtherSound® network must be synchronous with the Vi console's audio clock. This is ensured either by using the EtherSound® card as clock source of the EtherSound® network, or by feeding the device that is actually the EtherSound® network clock source with a word clock synchronous with the Vi console's audio clock.
The card is available for both Local Rack and Stagebox.

Current consumption (5 V)	750 mA max.
Operating temperature	0...40° C



* For more information on network topology and possible connections, please refer to the Ethersound documentation (www.ethersound.com).

** GPIs are GPOs on the Ethersound network, and vice versa. Audio outputs are audio inputs on the Ethersound network, and vice versa.



LEDs: **ES-100, ES-GIGA**
ES CLOCK

Indicate the mode selected with jumper J22.
Green: The card is the clock source of the EtherSound® network.
Red *(only in case of a ring network topology):* The card was defined to be the clock source of the EtherSound® network, but it is not, due to a device or cable failure in the ring.
Flashing red *(only in case of a ring network topology)**:*** The card was not defined to be the clock source of the EtherSound® network, but it actually is, due to a device or cable failure in the ring located just next to the card.
Dark: The card is not the EtherSound® clock source.

DIP Switch: **SW1**

DIP switch for D21m channel count setting:

1	2	3	4	5	6	7	8	Number of Channels
OFF	OFF	OFF	OFF	-	-	-	-	0 inputs
OFF	OFF	OFF	ON	-	-	-	-	8 inputs
OFF	OFF	ON	OFF	-	-	-	-	16 inputs
OFF	OFF	ON	ON	-	-	-	-	24 inputs
OFF	ON	OFF	OFF	-	-	-	-	32 inputs
OFF	ON	OFF	ON	-	-	-	-	40 inputs
OFF	ON	ON	OFF	-	-	-	-	48 inputs
OFF	ON	ON	ON	-	-	-	-	56 inputs
ON	OFF	OFF	OFF	-	-	-	-	64 inputs *(factory default)*
ON : ON	OFF : ON	OFF : ON	ON : ON	- - -	- - -	- - -	- - -	*NOT ALLOWED*
-	-	-	-	OFF	OFF	OFF	OFF	0 outputs
-	-	-	-	OFF	OFF	OFF	ON	8 outputs
-	-	-	-	OFF	OFF	ON	OFF	16 outputs
-	-	-	-	OFF	OFF	ON	ON	24 outputs
-	-	-	-	OFF	ON	OFF	OFF	32 outputs
-	-	-	-	OFF	ON	OFF	ON	40 outputs
-	-	-	-	OFF	ON	ON	OFF	48 outputs
-	-	-	-	OFF	ON	ON	ON	56 outputs
-	-	-	-	ON	OFF	OFF	OFF	64 outputs *(factory default)*
- - -	- - -	- - -	- - -	ON : ON	OFF : ON	OFF : ON	ON : ON	*NOT ALLOWED*

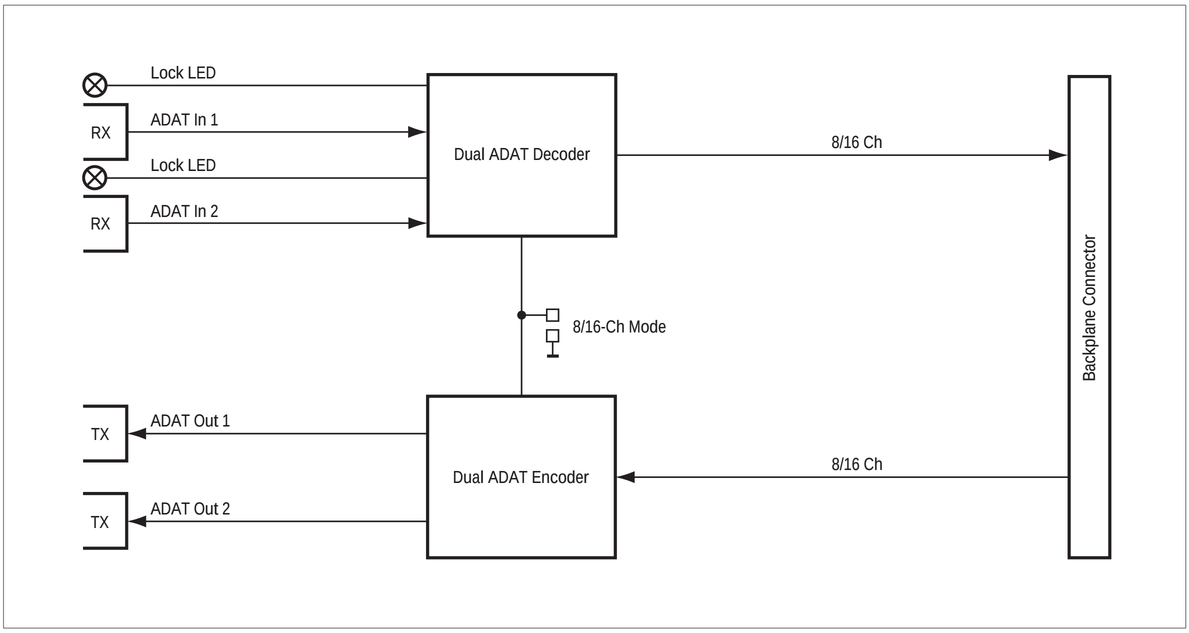
6.3.3 ADAT I/O Cards

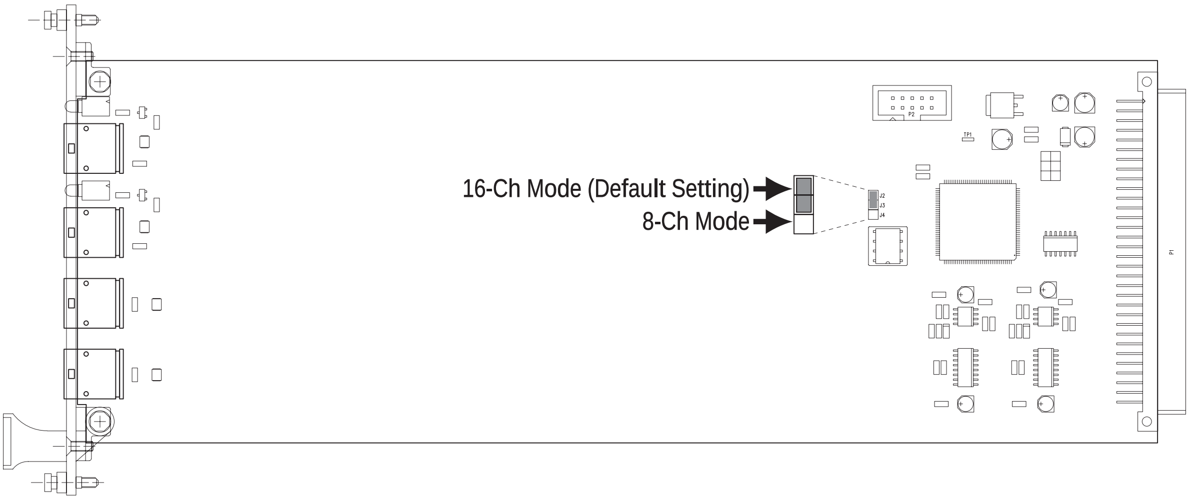
1.949.425, 1.949.429

Two optical eight-channel ADAT inputs and outputs. 48kHz operation. Optical inputs and outputs are provided on TosLink connectors.

The card is only available for the Local Rack.

Max. distance		5 m
Current consumption	(3.3 V)	0.1 A
	(5 V)	0.2 A
Operating temperature		0...40° C





LEDs:	**IN CH 1-8, 9-16**	These LEDs indicate that valid ADAT signals are available at the respective inputs.
Jumper:	**8/16 Ch Mode**	It is possible to restrict the number of channels from 16 to 8 using this jumper.

6.3.5 SDI Input Card

1.949.441

The HD/SD SDI (serial digital interface) 16-channel de-embedder card is able to de-embed eight or 16 audio channels from SDI-SD as well as from SDI-HD video streams. For the Vi I/O system it acts as an eight-or 16-channel audio input card. These two modes are determined by hardware switches located on the card.

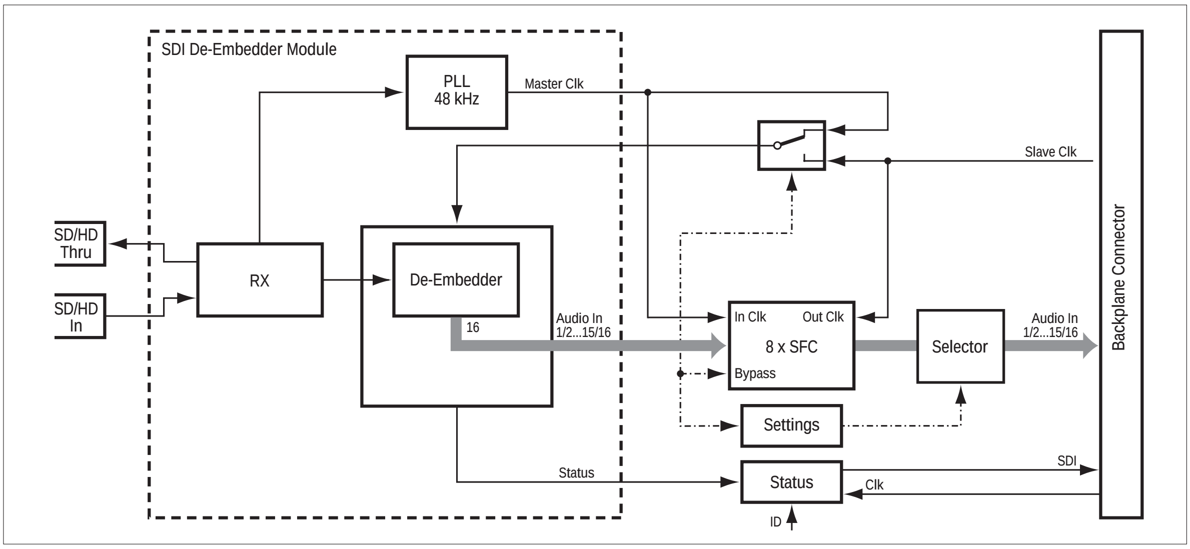
The SDI standard defines up to 16 audio channels transmitted within a video signal. These 16 channels are divided into four groups of four each. The user can determine by hardware switches whether all four groups, or only groups 1+2, or only groups 3+4 will be de-embedded.

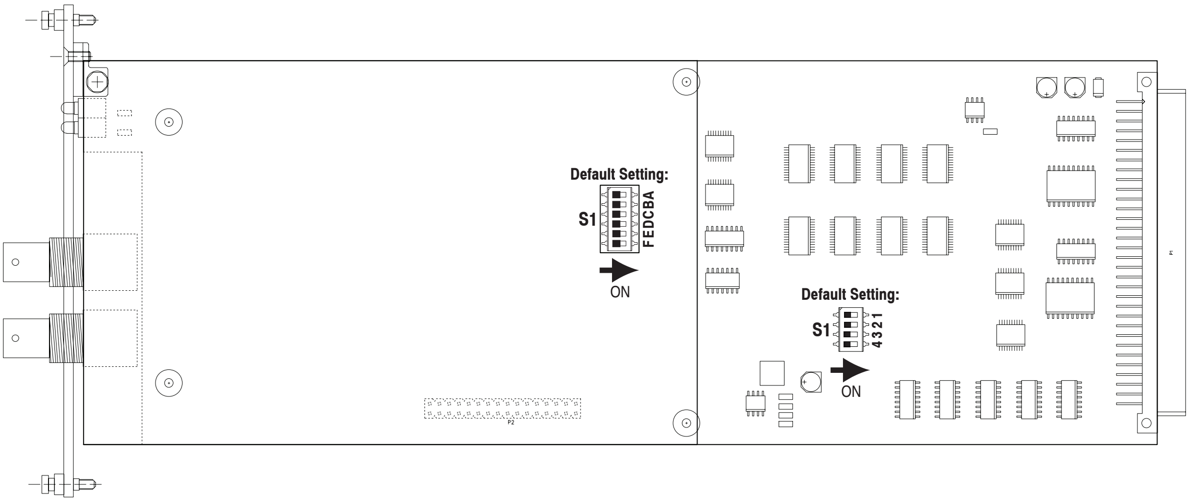
The card hosts SFCs (sampling frequency converters) that are bypassed per default. When bypassed, the SDI card is fully compatible to receiving embedded Dolby® E audio data. The SFCs can be enabled in case the audio extracted

Note: only available for the Local Rack.

ocal system. This means that the mixing console can run fully independent of the video sync used for SDI. *This card works at a sampling frequency of 48 kHz only*.

Modes	8- or 16-ch console input (de-embedder)
Selectable SDI groups	1&2, 3&4, or all
Video connectors	IN, THROUGH (BNC, 75 Ω)
Current consumption (5 V)	1 A
Operating temperature	0...40° C





LEDs: **SDI LOCK** Indicates a valid SDI signal at the input.

HD Indicates a valid HD SDI signal at the input.

DIP Switches: **S1**

Switch	Setting
1	OFF: 16-channel mode *(factory default)* ON: 8-channel mode
2	OFF: Group 1/2 used in 8-channel mode *(factory default)* ON: Group 3/4 used in 8-channel mode
3	OFF: SFC disabled *(factory default)* ON: SFC enabled
4	reserved *(must always be OFF; factory default)*

S1

Switch	Setting
A...F	reserved *(default: OFF)*

6.3.6 SDI I/O Card 1.949.442

The HD/SD SDI (serial digital interface) embedder/de-embedder card is able to handle video signals according to the SD as well as the HD standard. It can act as an eight-channel embedder, an eight-channel de-embedder, or as a combination of the two. Therefore, for the Vi I/O system it may act as an eight-channel audio input card, an eight-channel audio output card, or an eight-channel input and output card. These three modes are determined by hardware switches located on the card.

The SDI standard defines up to 16 audio channels transmitted within a video signal. These 16 channels are divided into four groups of four channels each. The user can select which two groups are to be embedded or de-embedded by hardware switches on the card: either groups 1&2, or groups 3&4. It is also possible to clear the SDI data structure possibly present in the incoming video signal and to allocate the groups from scratch.

The Vi SDI card hosts sampling frequency converters for both the audio inputs (de-embedding) and outputs (embedding). So the mixing console can run independent of the video sync used for SDI. The sampling frequency converters can be bypassed. When bypassed, the SDI card is fully compatible to transmitting the Dolby® E audio format.

This card works at a sampling frequency of 48 kHz only.

Note: only available for the Local Rack.

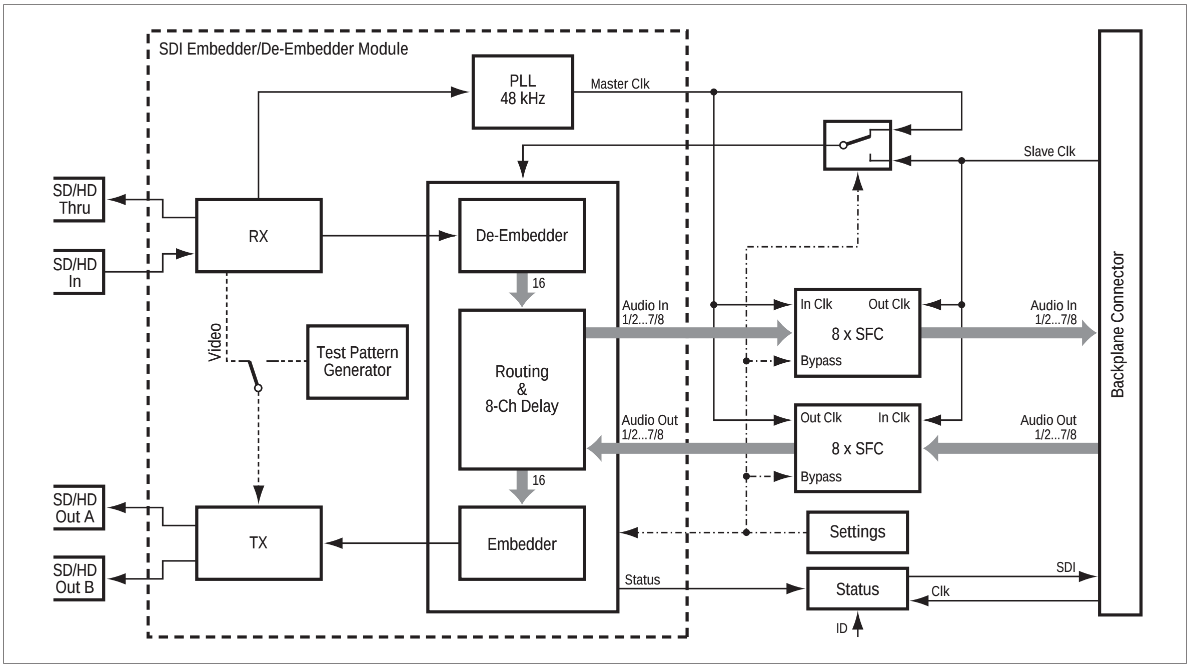
Modes 8-ch console output (embedder), 8-ch console input (de-embedder), or 8-ch console input and 8-ch console output (de-embedder/embedder)

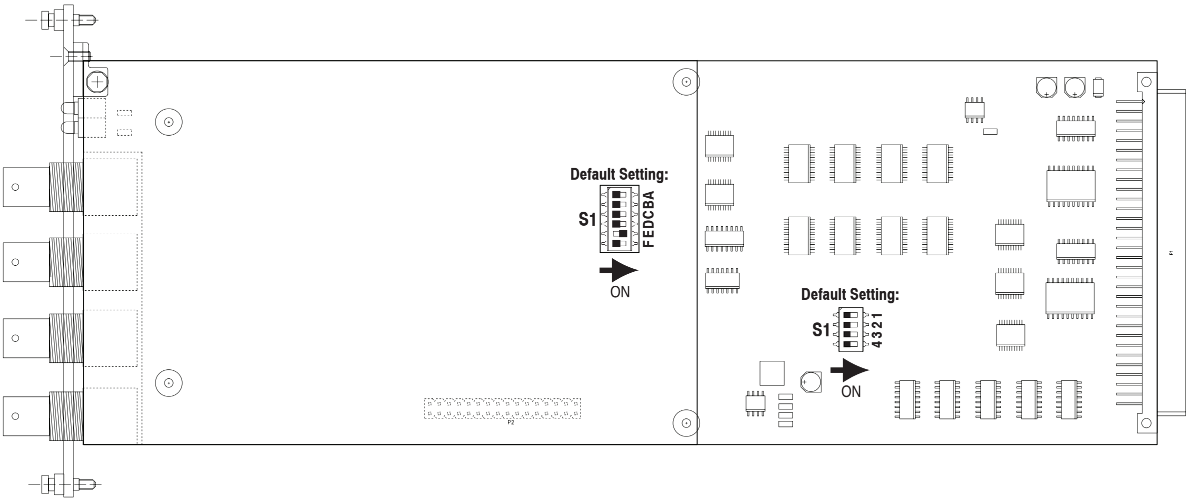
Selectable SDI groups 1&2, or 3&4

Video connectors IN, OUT A, OUT B, THROUGH (BNC, 75 Ω)

Current consumption (5 V) 1 A

Operating temperature 0...40° C





LEDs: **SDI LOCK** Indicates a valid SDI signal at the input.

HD Indicates a valid HD SDI signal at the input.

DIP Switches: **S1**

Switch	Setting
1	OFF: Enable de-embedder *(factory default)*
2	OFF: Enable embedder *(factory default)*
3	OFF: SFC bypass *(factory default)*
4	reserved *(must always be OFF)*

S1

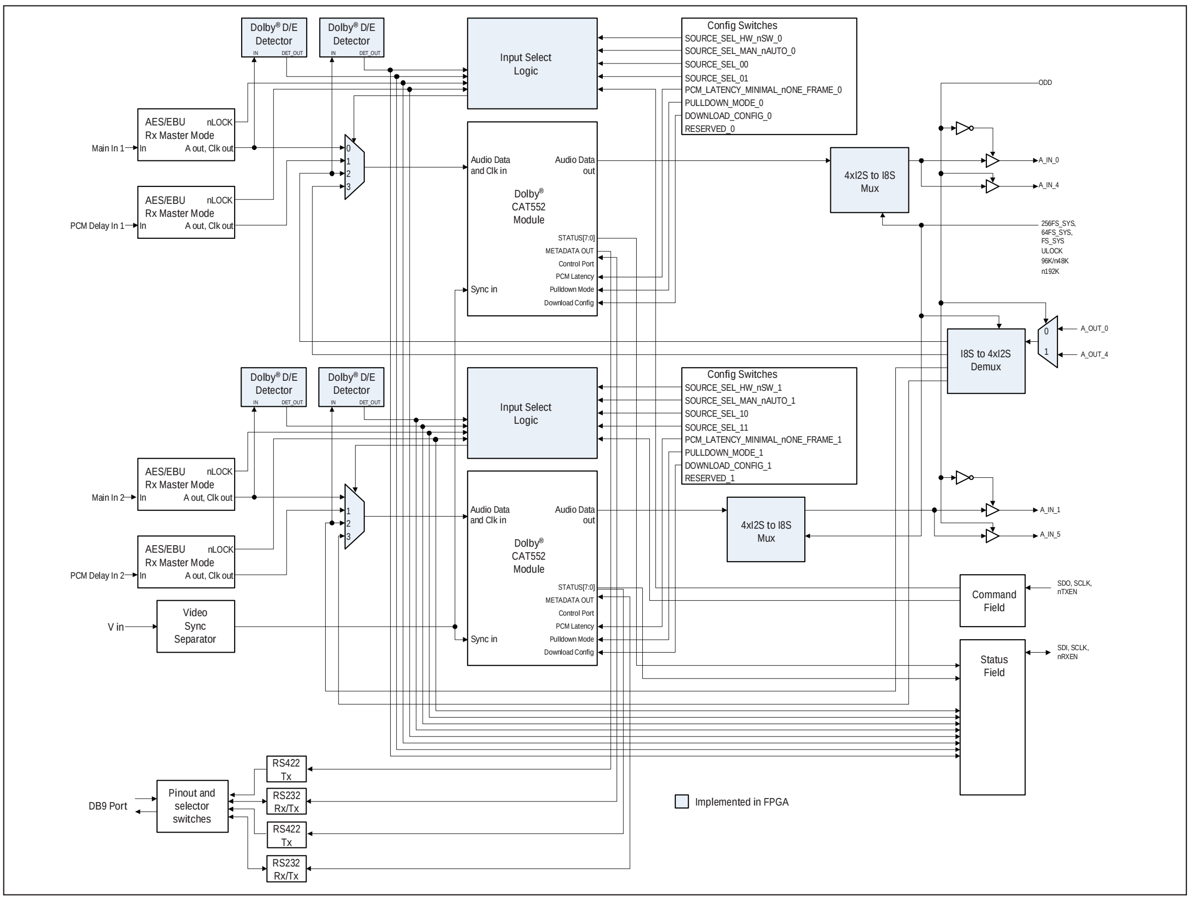
Switch	Setting
A	OFF: De-embedder groups 1&2 *(factory default)* ON: De-embedder groups 3&4
B	OFF: Embedder groups 1&2 *(factory default)* ON: Embedder groups 3&4
C	ON: All audio data in SDI will be cleared *(factory default: OFF)*
D	OFF: no delay *(factory default)* ON: 40 ms delay on all 8 SDI in channels
E	OFF: transparent for channel status bit ON: generate channel status bit *(factory default)*
F	OFF: NTSC 525 test pattern is generated if no SDI input signal is present *(factory default)* ON: NTSC 1080i60 test pattern if no SDI input signal is present

6.3.7 Dolby® E/Digital Decoder Card 1.949.443 (single-decoder) / 1.949.444 (dual-decoder)

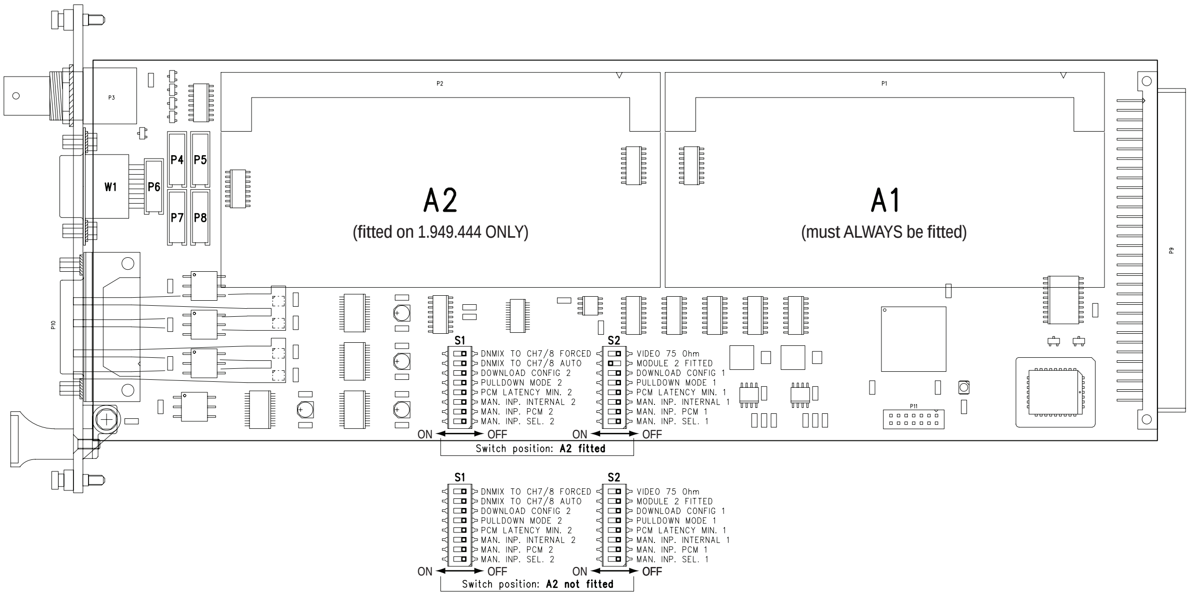
About Dolby® E Dolby® E allows encoding of up to 8 mono audio channels and some metadata into a pair of two channels (e.g. AES/EBU) by using 20 audio bits thereof. Both encoding and decoding processes create one video frame of delay. Since the encoded data is packaged in sizes of one video frame it is possible to "edit" the encoded stream, as long as the edits are synchronized with the video frames and the stream is not modified in any way (e.g. level changes applied). For more details on Dolby® E please refer to www.dolby.com.

The Decoder Card The Vi Dolby® E/Digital card hosts one or two Dolby® E decoder modules. Each one is functionally very similar to one Dolby® DP572 decoder. Both are operating independently, and the information given below is valid independently for both decoders as well. The dual-decoder card receives four AES/EBU pairs the front panel input, or eight mono channels from the console-internal patch (showing up as patch destinations). Each pair may contain a Dolby E or Dolby Digital encoded signal. The card returns a total of max. 16 channels to the console patch (showing up as patch sources).
The single-decoder card returns up to eight channels to the console patch (eight sources) and shows eight inputs on the patch. Input channels 5...8 are unused.

Notes: *The single-decoder card only works correctly if the Dolby® E decoder module is fitted in position A1.*
The card is only available for the Local Rack.



Current consumption (3.3 V) 0.2 A
(5 V) 0.8 A (1.949.443); 1.3 A (1.949.444)
Operating temperature 0...40° C



LEDs: **M1 / M2** Indicate that a valid AES/EBU signal is detected on main input 1/2.
P1 / P2 Indicate that a valid AES/EBU signal is detected on fallback input 1/2.
Note: These LEDs do not indicate Dolby® E status, but just the lock status of the AES/EBU inputs on the front panel.

DIP Switches: **S2.1 ... S2.3**

S2.1	S2.2	S2.3	Module 1 Input Select
x	x	OFF	Automatic source selection *(factory default: All OFF)*
OFF	OFF	ON	Front port main
OFF	ON	ON	Front port PCM delay
ON	OFF	ON	Rear (backplane / fallback) main
ON	ON	ON	Rear (backplane / fallback) main

While it is possible to manually select individual inputs both from the front panel connectors as well as from the console-internal patch, the card hosts an automatic source selection mode where the inputs are chosen automatically according to the following priorities:

- Whenever a valid AES/EBU signal is detected ("locked" status) on the 15-pin front panel connector, this input has priority over the console-internal patch sources. Hence if it is requested to feed the decoder with a console-internal signal selected via the patch window, no valid AES/EBU input signal is allowed on the front panel connector.
- However, if no valid AES/EBU signal is detected on the front panel inputs, the card is getting its inputs from the console-internal patch. These inputs are referred to as "Rear/Backplane Inputs". Selection is as follows:
  - Input 1, 2: Main priority input for Dolby® E signal, decoder 1.
  - Input 3, 4: Backplane input of decoder 1; is automatically selected in case no Dolby® E signal is present on main input (1, 2). Please note that a Dolby® E signal can be fed into this input, too, and it will be decoded correctly. However, if a Dolby® E signal is detected on the main input, this will be taken with higher priority.

S2.4

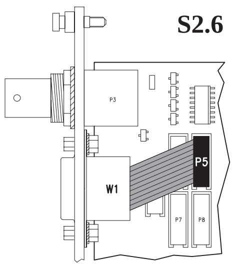
S2.4	PCM Latency (Module 1 only)
OFF	PCM signal is delayed by 1 video frame *(factory default)*
ON	PCM signal is minimally delayed

Decoding a Dolby® E stream always causes a delay of one video frame. In case a regular PCM signal is fed to the card, this can be delayed by one video frame, too. If required, this delay may be de-activated in order to pass through a PCM signal with a minimal delay. The front panel VIDEO IN sync input is used to detect video frames in order to delay the PCM signal accordingly. The video sync input doesn't necessarily have to be connected in case of Dolby® E , since the sync is indicated within the Dolby® E stream.

S2.5

S2.5	Module 1 Pulldown Mode
OFF	Pulldown mode is off *(factory default)*
ON	Pulldown mode is on

Pulldown mode ON allows the input of audio signals with a "drop frame" sampling frequency of 47.952 kHz instead of 48 kHz. The output, however, always runs at 48 kHz.

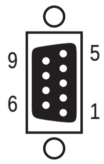


S2.6

S2.6	Module 1 Configuration Download
OFF	Standard operation *(factory default)*
ON	Configuration download via RS232

If firmware download to decoder module 1 is required, plug the short flat cable (W1) coming from the METADATA OUT front-panel socket to the PCB socket P5 (labeled UPDATE1).
The pin assignment of the METADATA OUT socket (9-pin D-type, female) in this case is as follows:

Solder/Crimp View
(or Socket View)



Pin	Signal	Pin	Signal
1	n.c.	6	n.c.
2	DOUT_1	7	n.c.
3	DIN_1	8	n.c.
4	n.c.	9	n.c.
5	n.c.		

S2.7

S2.7	Module 2 Installed
OFF	No *(factory default if not installed, i.e., for 1.949.443)*
ON	Yes *(factory default if installed, i.e., for 1.949.444)*

S2.8

S2.8	Video Termination
OFF	Hi-Z *(factory default)*
ON	75 Ω

S1.1 ... S1.3

S1.1	S1.2	S1.3	Module 2 Input Select
x	x	OFF	Automatic source selection *(factory default: All OFF)*
OFF	OFF	ON	Front port main
OFF	ON	ON	Front port PCM delay
ON	OFF	ON	Rear (backplane) main
ON	ON	ON	Rear (backplane) PCM delay

Same as S2.1 ... S2.3 above, but for module 2 (if installed).

S1.4

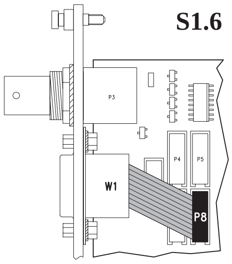
S1.4	PCM Latency (Module 2 only)
OFF	PCM signal is delayed by 1 video frame *(factory default)*
ON	PCM signal is minimally delayed

Same as S2.4 above, but for module 2.

S1.5

S1.5	Module 2 Pulldown Mode
OFF	Pulldown mode is off *(factory default)*
ON	Pulldown mode is on

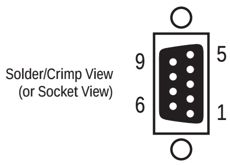
Same as S2.5 above, but for module 2.



S1.6

S1.6	Module 2 Configuration Download
OFF	Standard operation *(factory default)*
ON	Configuration download via RS232

If firmware download to decoder module 2 is required, plug the short flat cable (W1) coming from the METADATA OUT front-panel socket to the PCB socket P8 (labeled UPDATE2).
The pin assignment of the METADATA OUT socket (9-pin D-type, female) in this case is as follows:



Pin	Signal	Pin	Signal
1	n.c.	6	n.c.
2	DOUT_2	7	n.c.
3	DIN_2	8	n.c.
4	n.c.	9	n.c.
5	n.c.		

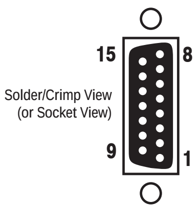
S1.7 / S1.8

S1.7	S1.8	Downmix to Ch 7/8 (or 15/16, resp.)
OFF	OFF	No downmix *(factory default)*
ON	OFF	Automatic downmix
OFF	ON	Forced downmix

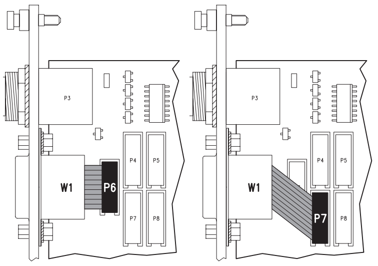
Metadata and Downmixing: A Dolby® E stream contains metadata with various information on the encoded signal. This information can be read out from the front panel connector. The Vi Dolby® E decoder card only uses this information in case a 2-channel stereo downmix is required from a 5.1-channel surround signal within the Dolby® E stream; then the decoder interprets the center and surround channel levels and uses them for the internal downmixer that is activated by the DIP switches S1.7 and S1.8. The downmix can be made constantly available and, subsequently, overwriting any audio data that was contained on these channels beforehand ("forced downmix"), or it is possible to "fill" the channels 7/8 or 15/16 only if the metadata indicate that these channels are not being used otherwise (automatic downmix).

Connector Pin Assignments:

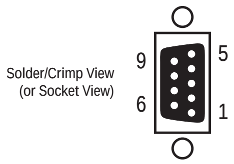
2 x AES IN MAIN/PCM (15-pin D-type, female)



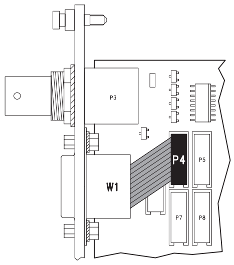
Pin	Signal	Pin	Signal
1	Main In 1 +	9	Main In 1 –
2	Main In 1 Chassis	10	PCM Delay In 1 Chassis
3	PCM Delay In 1 –	11	PCM Delay In 1 +
4	n.c.	12	n.c.
5	Main In 2 +	13	Main In 2 –
6	Main In 2 Chassis	14	PCM Delay In 2 Chassis
7	PCM Delay In 2 –	15	PCM Delay In 2 +
8	n.c.		



METADATA OUT (9-pin D-type, female)
The Metadata Out socket allows sending the meta data of either module or of both modules at once.
If the meta data of either decoder module 1 or 2 is required, plug the short flat cable (W1) coming from the METADATA OUT front-panel socket to the PCB socket P6 (labeled META1; *factory default*), or to PCB socket P7 (META2), respectively.
The pin assignment of the METADATA OUT socket (9-pin D-type, female) in this case is as follows:

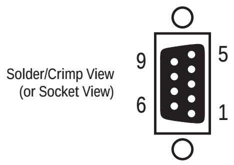


Pin	Signal	Pin	Signal
1	Chassis	6	GND
2	n.c.	7	n.c.
3	META_1+ / META_2+	8	META_1– / META_2–
4	GND	9	Chassis
5	n.c.		



If the meta data of both decoder modules is required, plug the short flat cable (W1) coming from the METADATA OUT front-panel socket to the PCB socket P4 (labeled META1+2).

Please note that in this case the pin assignment of the METADATA OUT socket (9-pin D-type, female) is *non-standard*:

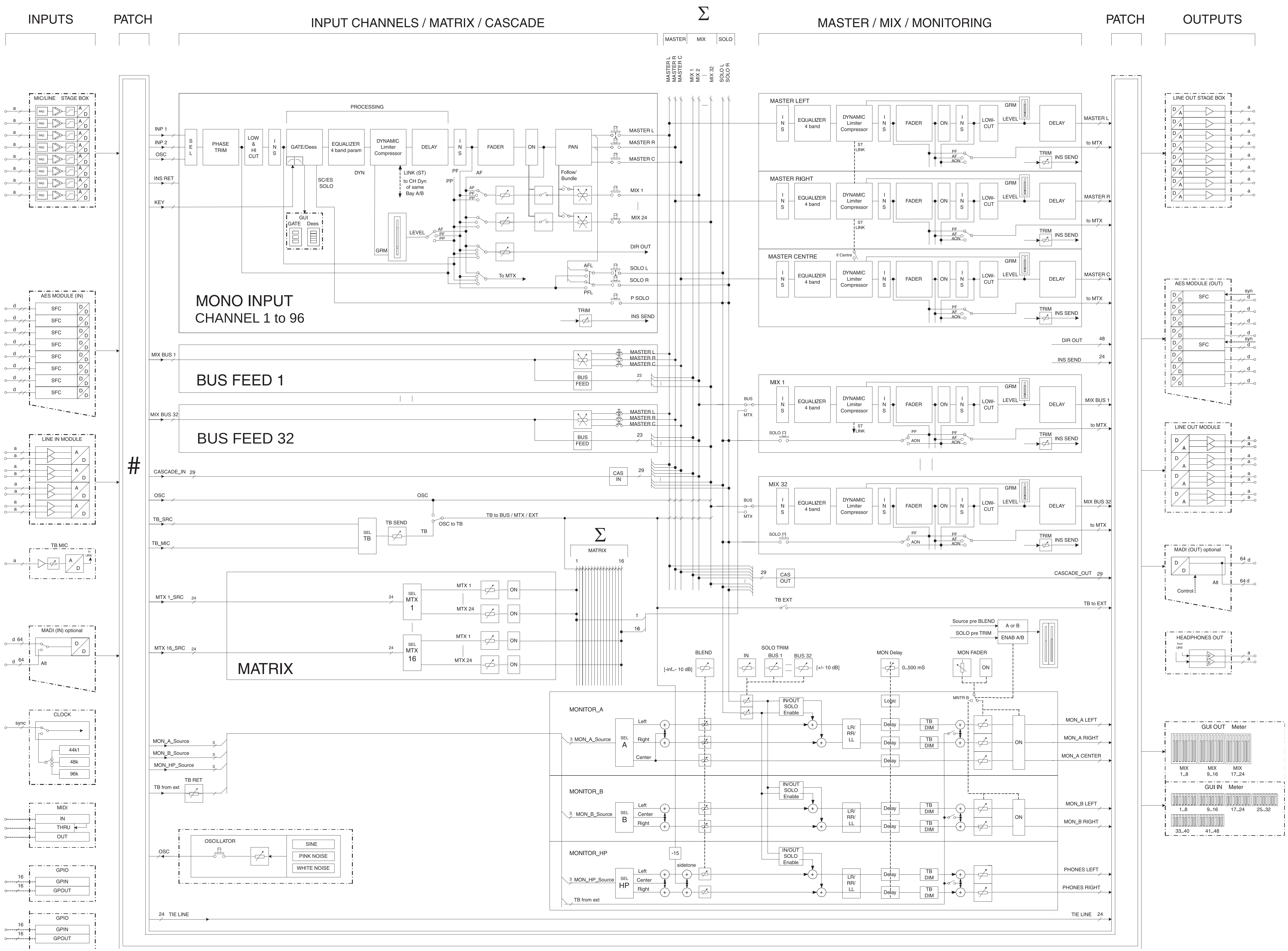


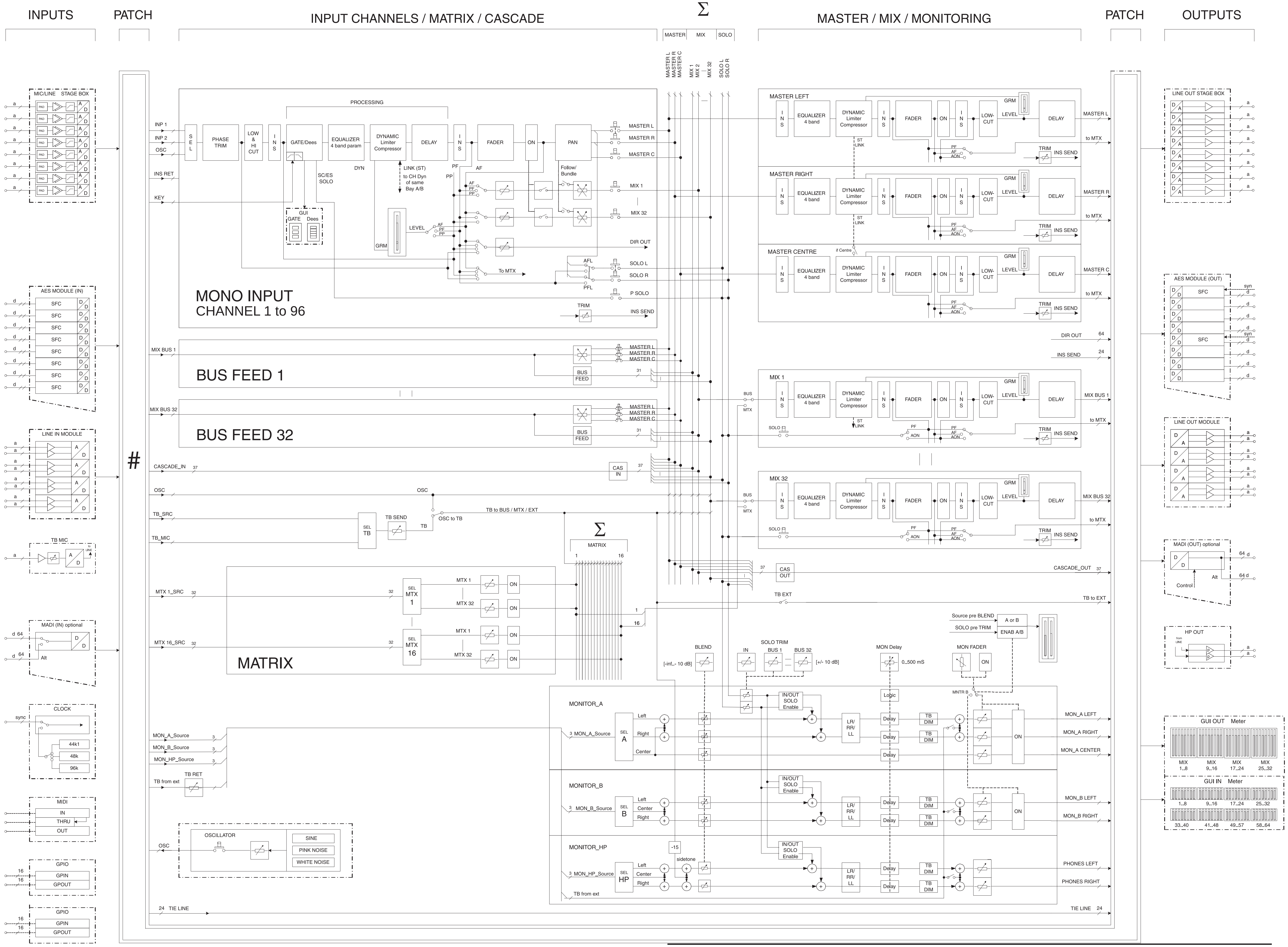
Pin	Signal	Pin	Signal
1	Chassis	6	GND
2	n.c.	7	META_2–
3	META_1+	8	META_1–
4	META_2+	9	Chassis
5	n.c.		

Possible Pitfalls with Dolby® E

In order to transmit or record a Dolby® E encoded signal, *the whole signal path must be 100% transparent*, regarding the 20 audio bits contained within the data stream. In case of problems with decoding the Dolby® E signal and possibly getting white noise instead of the decoded signal, the whole signal path should be checked. It may be worthwhile verifying the following points:

- Are there any sampling frequency converters (e.g. when using the D21m Dolby® E decoder card together with the Vi SDI card) in the signal chain? If so, they must be bypassed; otherwise the Dolby® E stream is modified and cannot be decoded anymore.
- In case the signal is sourced from a video tape machine: Is the machine set up to be transparent for the recorded audio signals? Several machines require seting the tracks to “DATA” mode in order to guarantee unity gain while recording or playing back Dolby® E streams.
- Is the card receiving the Dolby® E stream from the console-internal patch? If so, are both tracks patched to the correct two inputs of the card? (Decoder 1 main: channels 1 and 2; decoder 1 PCM: channels 3 and 4; decoder 2 main: channels 5 and 6; decoder 2 PCM: channels 7 and 8).
- If getting a wrong signal or no signal at all: Are any AES/EBUsignals present at the front panel while console-internal streams should be decoded? If the card is in “automatic source selection” mode, the front inputs have top priority, regardless whether a Dolby® E stream is recognized or not.


INPUTS
PATCH
INPUT CHANNELS / MATRIX / CASCADE
Σ
MASTER
MIX
SOLO
MASTER / MIX / MONITORING
PATCH
OUTPUTS
MIC/LINE STAGE BOX
AES MODULE (IN)
LINE IN MODULE
TB MIC
MADI (IN) optional
CLOCK
MIDI
GPIO
#
PROCESSING
PHASE TRIM
LOW & HI CUT
GATE/Dees
EQUALIZER 4 band param
DYNAMIC Limiter Compressor
DELAY
FADER
ON
PAN
MONO INPUT
CHANNEL 1 to 96
BUS FEED 1
BUS FEED 32
MATRIX
CASCADE_IN
TB_SRC
TB_MIC
MTX 1_SRC
MTX 16_SRC
MON_A_Source
MON_B_Source
MON_HP_Source
TB from ext
OSCILLATOR
SINE
PINK NOISE
WHITE NOISE
TIE LINE
MASTER LEFT
MASTER RIGHT
MASTER CENTRE
MIX 1
MIX 32
CASCADE_OUT
TB to EXT
MONITOR_A
MONITOR_B
MONITOR_HP
MON_A LEFT
MON_A RIGHT
MON_A CENTER
MON_B LEFT
MON_B RIGHT
PHONES LEFT
PHONES RIGHT
LINE OUT STAGE BOX
AES MODULE (OUT)
LINE OUT MODULE
MADI (OUT) optional
HEADPHONES OUT
GUI OUT Meter
GUI IN Meter


INPUTS
PATCH
INPUT CHANNELS / MATRIX / CASCADE
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SOLO
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FADER
ON
PAN
MONO INPUT
CHANNEL 1 to 96
BUS FEED 1
BUS FEED 32
CASCADE_IN
MATRIX
OSCILLATOR
SINE
PINK NOISE
WHITE NOISE
MASTER LEFT
MASTER RIGHT
MASTER CENTRE
MIX 1
MIX 32
MONITOR_A
MONITOR_B
MONITOR_HP
LINE OUT STAGE BOX
AES MODULE (OUT)
LINE OUT MODULE
MADI (OUT) optional
HP OUT
GUI OUT Meter
GUI IN Meter
TIE LINE

Introduction to VM^2

VM^2 is a unique new patented feature, which allows the status information for AKG wireless microphones to be displayed directly within the channel strip that they are connected to. This feature makes further use of Harman's HiQnet network control protocol and brings many benefits including streamlining of the workflow and increase in speed of problem diagnosis.

VM^2 Concepts

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 AKG Wireless Microphone system, and a Soundcraft Vi Series console.

What is 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 System Architect provides a master control and monitoring program that allows all equipment on the HiQnet network to be controlled from a single user interface.

How is HiQnet used on the Vi-series?

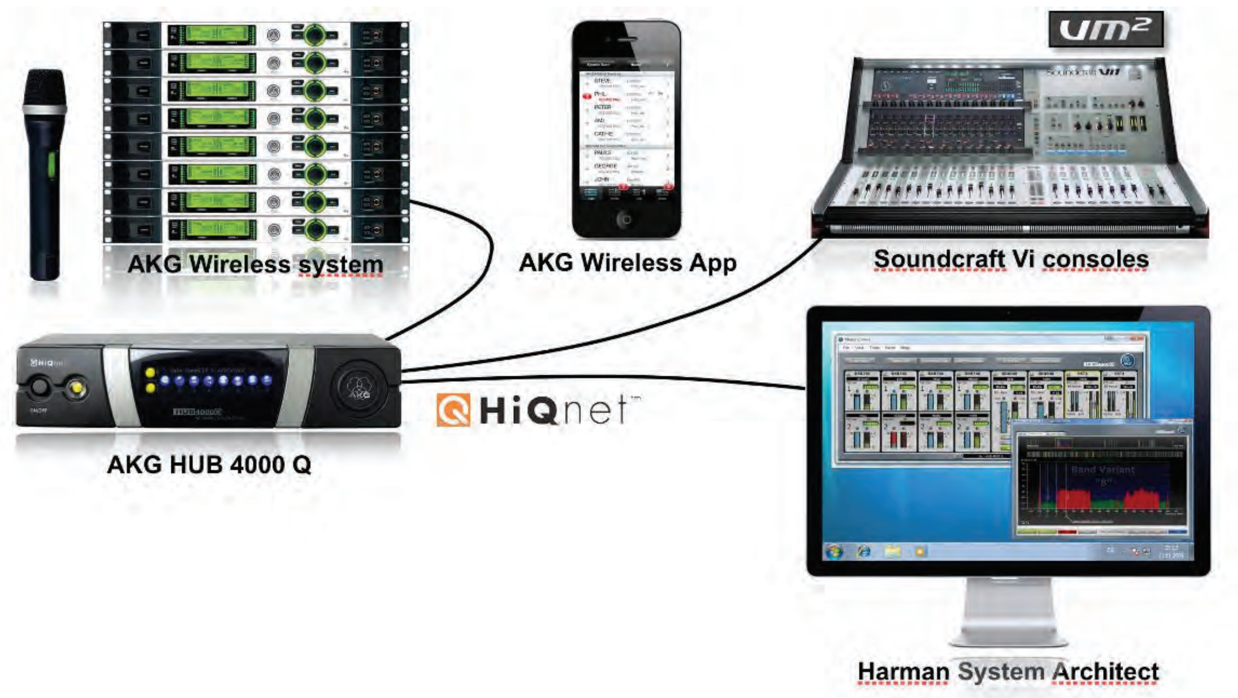
Since V3.0 software, the Vi-series has included only a limited HiQnet functionality which allows error messages generated by other Harman devices to be displayed in the console's message log – for example thermal overload messages generated by Crown amplifiers. In addition the consoles were able to transmit a Venue Preset recall - the HiQnet equivalent of a MIDI Program Change – to all other devices in the network. Version 4.5 now adds significantly to this existing HiQnet functionality with the ability to monitor wireless microphone status within the console GUI.

HiQnet is also used to used to provide remote control of console parameters, for example via the Soundcraft ViSi Remote iPad® app. See chapter 26.

How does VM² work?

In our Microphone Monitoring setup, the AKG wireless receivers are connected to an AKG device called Hub4000Q, which receives data cables from up to 8 wireless receiver units and provides a single HiQnet Ethernet connection through which the monitoring information for the 8 microphones can be accessed by other devices.

In the simplest configuration, one Hub4000Q can be connected directly to the HiQnet port of a Vi Series console, and the monitoring functionality will be provided on the console. Typically however other devices will be required to share the HiQnet data from the microphones, and in this case an Ethernet switch (not shown in the diagram below) will be used to allow other Harman devices and/or a wired or wireless computer running Harman's System Architect software to be connected into the network.



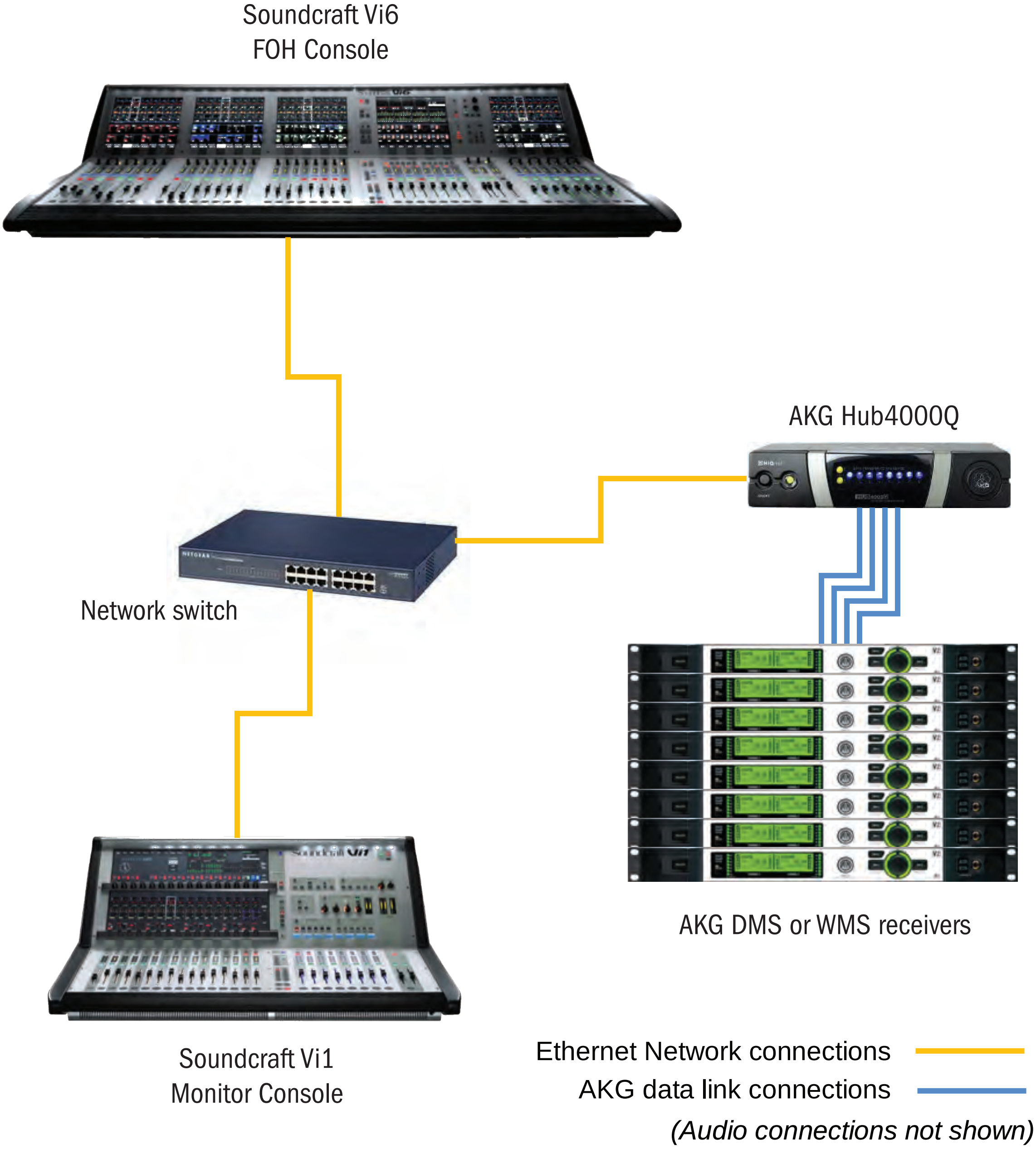
Setting up a VM² Network

In the following setup information, a basic understanding of computer network configuration is assumed.

Basic Network Topology

The example illustrated shows a typical setup with a front-of-house and Monitor console and a rack of AKG wireless receivers.

If only one console is being used, and there is no other HiQnet-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 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.

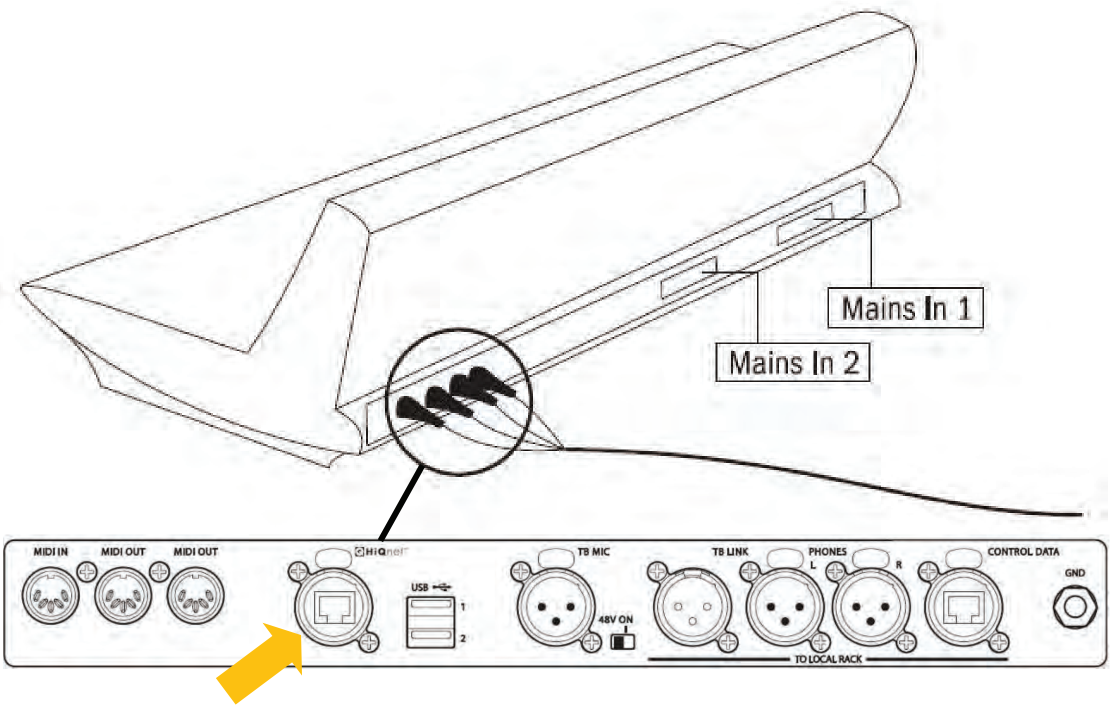


Identifying the HiQnet Network connector on Vi consoles

Vi1
The HiQnet port is on the rear of the control surface, adjacent to the rear USB sockets.

Important! The HiQnet port on early Vi1 consoles (serial numbers lower than 30210956) is only able to drive short cable lengths, max recommended length 1 metre (3 feet). Therefore use a network switch positioned close to the rear of the console to extend the cable length if required.

Vi2/Vi4 /Vi6
The HiQnet port is located on the rear of the Control Surface, adjacent to the MIDI ports. The port is capable of driving standard Ethernet distances of up to 100m.



Detailed Console Operation of VM^2

Setting up and using VM^2 monitoring on a Vi Series console falls into four simple steps:

- Connect up the HiQnet network as described in the previous section and check that the IP address configuration is correct on all devices.

- Connect up the audio connections from the AKG Receivers to the console (either to Stagebox or local inputs as appropriate).

- Associate each AKG receiver with a console connector using the mapping table in the Vi's HiQnet Setup page.

- The microphone monitoring information will now automatically be displayed on any channel strip which uses an AKG-mapped physical input connector as its patched source.

The following pages describe in more detail how to carry out the second two steps. Note: Screenshots shown are for Vi2/4/6. Vi1 will have same controls but layout will differ).

Using the Console's HiQnet Setup page to set up an IP configuration

Press the MENU button, select the System tab, then the HiQnet tab to access the Setup page.

Configuring the IP Address

Ensure the console's HiQnet port is connected to the HiQnet network, and use the IP Config and IP address controls to set up a valid IP address for the console.

If you wish to set the IP address and subnet manually, set the IP CONFIG Vistonics control to 'MAN'. Use the IP ADDRESS and SUBNET MASK Vistonics controls to enter a valid configuration and then press the 'SET' button within the IP CONFIG control.

The available ranges of valid IP addresses are listed below:
Note: 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.

10.0.0.0 - 10.255.255.255
172.16.0.0 - 172.31.255.255
192.168.0.0 - 192.168.255.255

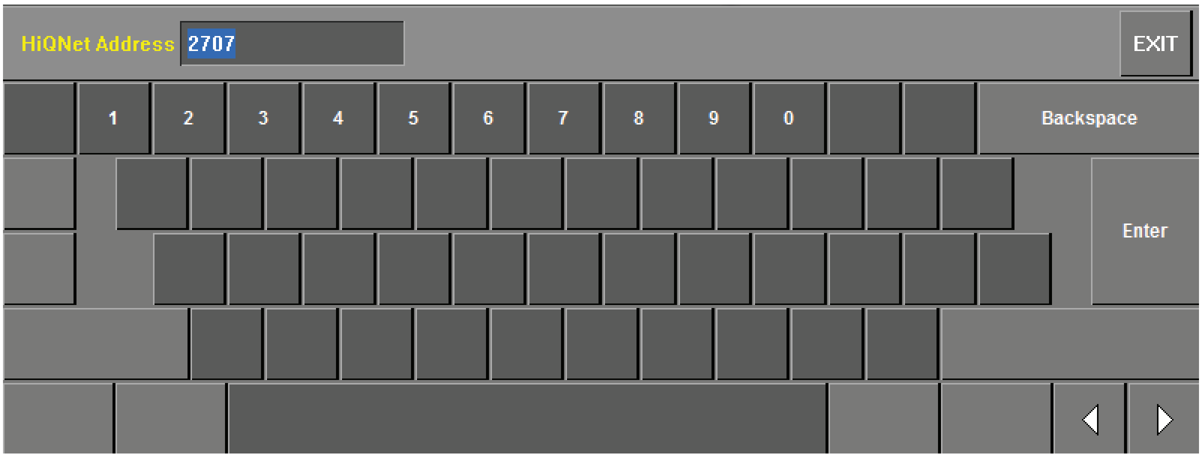
If you are using a DHCP server to automatically configure the IP setup, set the IP CONFIG Vistonics control to 'DHCP'. Wait for several seconds until an IP address appears in the IP ADDRESS Vistonics control fields. When a valid address is established you will also see it on the far left of the HiQnet page.

At this stage, all the controls in the HiQnet Vistonics section will be greyed out, indicating that HiQnet is disabled.

Setting the HiQnet Node Address

Before enabling HiQnet, use the HIQNET ADDR Vistonics control to set a suitable HiQnet node address for the console. 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 will need to use System Architect to find out what the node addresses of other equipment are to ensure there are no conflicts. The Vi node address will be set to a default value of 2717 unless it has been changed previously, and may be left at this address as long as this is not already being used elsewhere in the network.



To set the HiQnet address, press the button within the HiQnet Address Vistonics control (see screenshot on previous page). A keyboard will be opened which will allow the default address to be edited.

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.

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.

Using the Device List

Device List

The Device list occupies the centre of the HiQnet Setup page, and will initially be completely empty. Once HiQnet has been turned on, and a HiQnet network is attached to the console, the console will search for any attached AKG 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 the mics.

The DEVICE column shows the names of all of the AKG microphones that have been discovered on the network. This column is automatically populated.

The NET column in the list shows the HiQnet node address of the AKG Hub4000Q to which each microphone is connected. In the example shown above there is only one Hub4000Q connected, but it is possible for multiple Hub4000Qs 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 VM^2 monitoring information, it is necessary to 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 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.

Associating the Microphones with Audio Connectors

Use the MAP DEVICES Vistonics control to scroll the list and select a microphone, then press the AUDIO PATCH button to open the patching matrix.

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 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.

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.

When all the assignments have been made, close the Audio Patch page by pressing EXIT button on the screen, or pressing the Vistonics button in the AUDIO PATCH control field.

Returning to the Device List, you should now have a list of devices with all three columns indicating information in this format:

AKG Device name

Hub4000Q node address

Audio Patch assignment

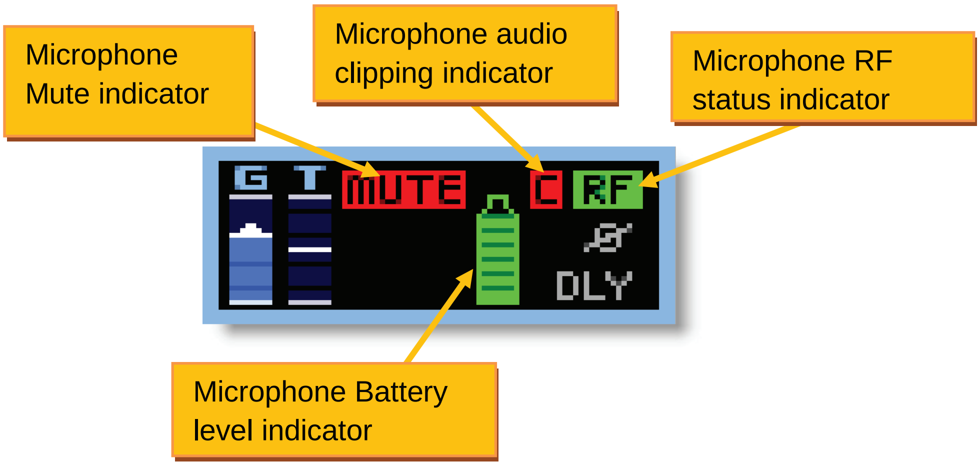
Using the Monitoring Information Display on the Channel Strips

Once the Device list has been populated with discovered Microphone devices, and audio patch connectors have been assigned to these, the Monitoring information will automatically be displayed in the channel strips where those audio patches are used.
The screenshot below has the first three channels with AKG mics assigned, and shows the location of the AKG Monitoring info.

The Monitoring elements initially appear as mini-icons within the Input section at the top of the Channel Strip, as shown in this screenshot:

Note that the names of the AKG microphones appear in the channel label display at the bottom of the channel strip.

In more detail, the elements that are added to the input channel overview are as follows:



The displayed Microphone monitoring elements shown on previous page provide the following functions:

Microphone Mute status indicator
Shows red 'MUTE' icon if the AKG mic is switched to muted state.
When unmuted, this indicator icon disappears.

Microphone Audio clipping indicator
A red 'C' is displayed momentarily if the audio within the wireless mic audio path reaches full scale.

Microphone RF status indicator
A red or green 'RF' icon is displayed to indicate the health of the AKG 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. Note: If the microphone transmitter is switched off, the RF indicator will also change to red, as the system is unable to tell the reason for the lack of RF.

The actual value in dB of the RF signal strength is displayed in the Vistonics detailed monitoring information display (see next page).

Microphone Battery level indicator
A red or green battery level icon indicates the health of the battery within the AKG wireless transmitter. The interior of the battery icon contains 7 segments showing varying battery level. The colour coding of the battery icon is as follows:

Green	Segments 3-7	35-100% remaining
Amber	Segment 2	25% remaining
Red	Segments 0-1	1 hour remaining

The number of hours remaining in the transmitter battery is displayed in the Vistonics detailed monitoring information display (see next page) and varies with the type of battery used.

Network Error display

The microphone status indications are only valid if they are being transmitted by a HiQnet network therefore it is useful to know whether there are network problems. The system is able to differentiate between a network error/disconnection and for example out of range RF or switched off transmitter. In the case of a network error, the AKG elements only within the input strip overview on all channels will change to a greyed-out condition as shown below:

If the network is disconnected, or there is a network connection error, the following diagnostic indicators are also available:

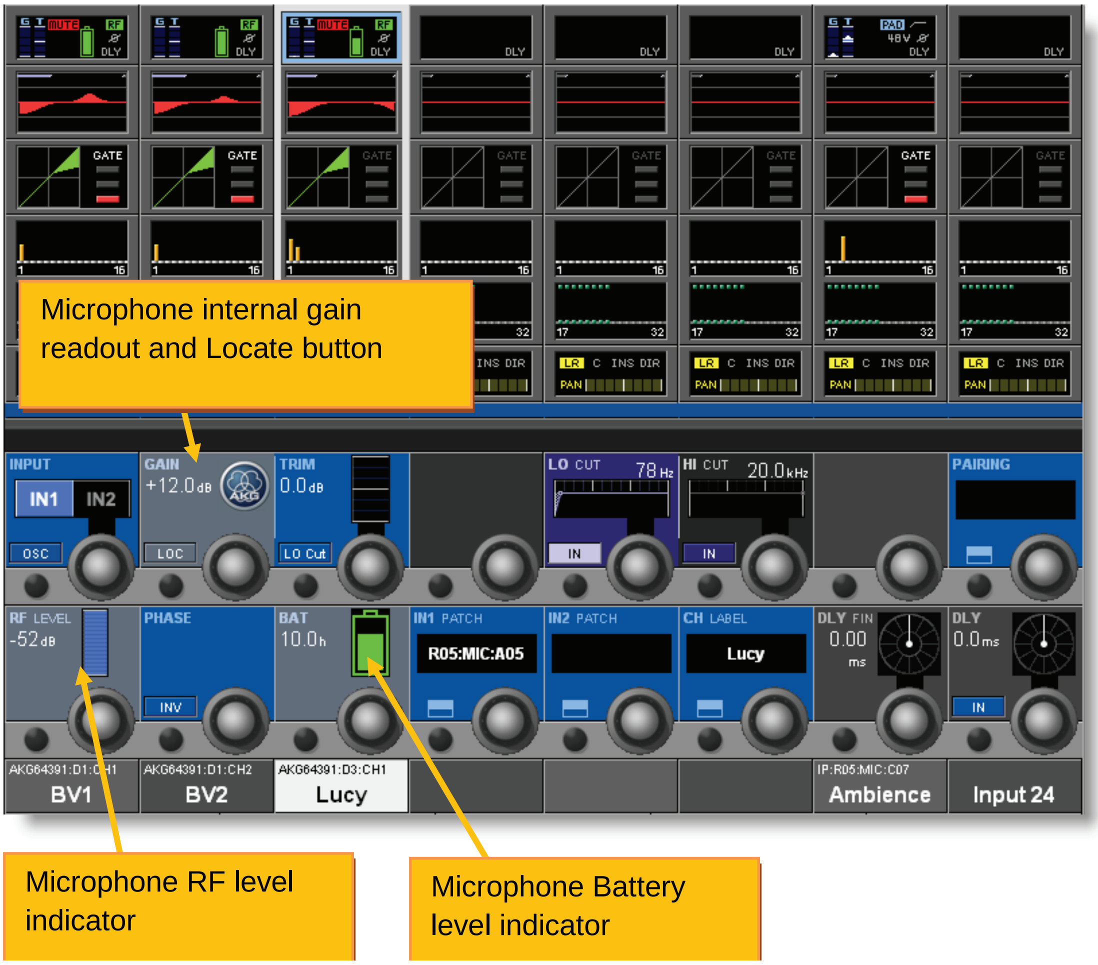
· AKG devices in the Device List in the HiQnet Setup page will also appear greyed out.
· If the network is disconnected between the console and the router, the HiQnet icon in the main console diagnostic display area will change from green to grey.

Vistonics Detailed Monitoring Display

To see more detailed information about the AKG 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 AKG-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 this information is display-only, there is no control possible over the AKG microphones, other than the Locate button. Note also that the input gain control of the Vi microphone input preamp will be automatically set to 0dB gain and is not available for control on channels that have AKG microphones assigned.



The display of internal Gain of the AKG microphone differs according to the type of microphone system being monitored (Note: this cannot be controlled from the console – it is a display of the microphone's internal parameter only!)

- *On WMS4500 systems: Displays the current value of the user-adjustable audio gain of the transmitter.*
- *On DMS700 systems: Displays the output trim level of the receiver.*

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.

TROUBLESHOOTING A VM² SETUP

Problem
HiQnet cannot be switched ON in the Menu-System-HiQnet page because there is no IP address.

Solution
1. Check the cable connection to the HiQnet port on the console – if there is no valid Ethernet connection, it will not be possible to switch on the HiQnet functionality.

2. Assuming cable connection is OK, if DHCP mode is selected on the console, check that the router that is being used has a DHCP server capability and that this is enabled in the router setup.

3. It is possible to connect the console directly to the Hub4000 without a router, with the console set to DHCP mode. In this case there will be a considerable delay (up to 60 secs) before an IP address is allocated, as this has to be negotiated between the console and the Hub.

4. If using a manually set address, the console's IP address must have the same subnet mask as the Hub4000Q, so System Architect must be used to discover what that is. Once the address is known, set the console to a different address but with the same subnet mask.

Problem
No AKG receivers are detected by the console – the list on the HiQnet page is empty

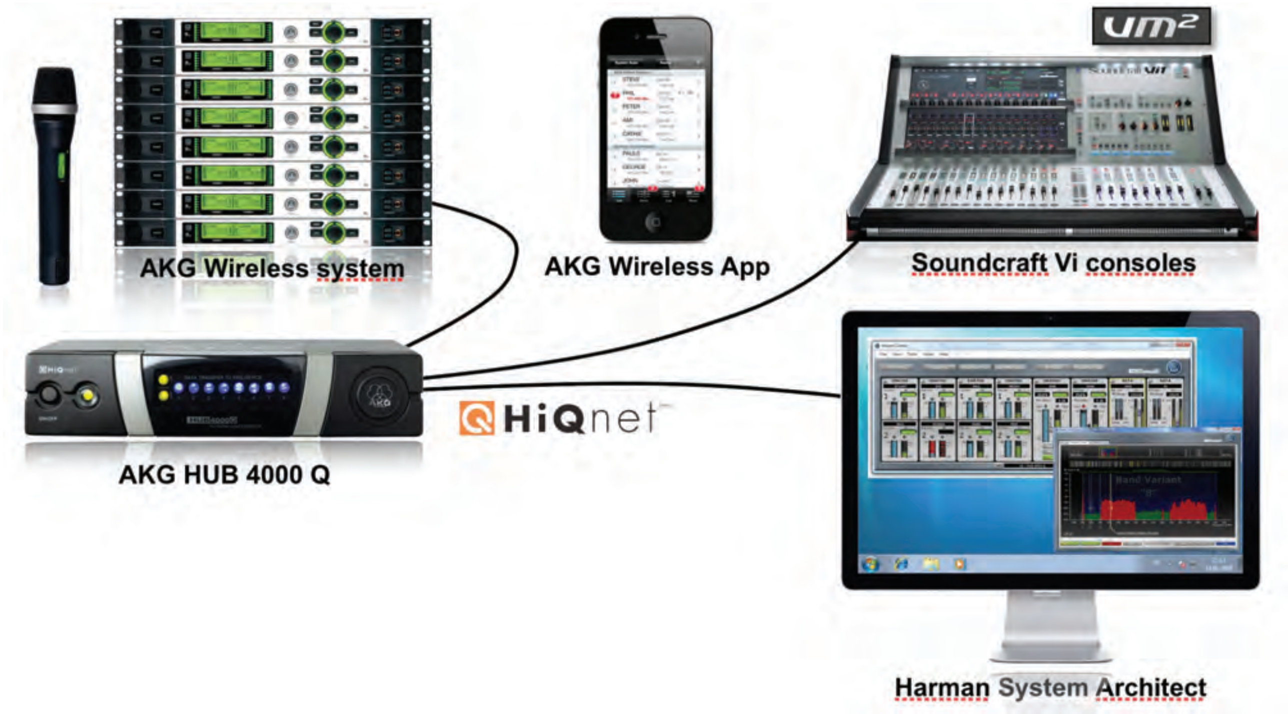
Solution
1.Check that all AKG receivers are connected to the Hub4000Q, the Hub4000Q front panel LEDs show connection status of the attached receivers, and there is a network connection between the Hub4000Q and the console (via a network switch if necessary)

2. Check that the IP Config of the console and the Hub4000Q match – if using a router with DHCP server and you do not want to use a manually set IP address, make sure the console is set to 'DHCP' mode in the Menu-System-HiQnet page.

Problem
The AKG receivers are listed in the Device list on the HiQnet page are greyed out.

Solution
This indicates that there is a network connection has previously existed correctly but there is now a connection error – either the Hub4000Q has been disconnected or switched off, or there is another connection problem between the Hub and desk or router and desk. Check all network connections.

FAQs - AKG Wireless Systems



HiQnet® Remote Control and Setup

These Frequently Asked Questions (FAQs) may help if there are issues with working or setting up the HUB 4000 Q.

The FAQs help you to get started quickly with setting up and remote control of your AKG wireless system with the PC software Harman System Architect™, the AKG Wireless iPhone® App and Soundcraft Vi consoles VM^2 feature based on Harman HiQnet® protocol using the HUB 4000 Q.

If you have further questions check out the quickstart guide and manuals which can be found online at http://www.akg.com/hiqnet.

If you need further help please send an email to hiqnet@akg.com.

FAQ 1: How do I setup my HUB 4000 Q(s) and my wireless system together with System Architect/AKG Wireless iPhone App?

See Quick Start Guide steps 1 - 3 and FAQs.

FAQ 2-1: System Architect starts up and no HUB 4000Q is detected!

If the HUB 4000 Q is not showing up at HiQnet Explorer, System Architect has no network connection to the HUB 4000 Q.

Possible reasons:

- *The HUB is physically not connected to the HiQnet network*
  → check your ethernet/network cables and check if your router and/or switches are powered on
  → If you connect the HUB 4000 Q directly to your PC than you have to use a crossover network cable

- *The HUB 4000 Q is not powered on*
  → check the power connection of the HUB 4000 Q

- *The HUB 4000 Q has no or wrong IP address/subnet settings*
  → See FAQ 3

FAQ 2-2: AKG Wireless iPhone App shows no list entries/devices at it‘s main list screen!

→ See FAQ 2-1

FAQ 2-3: Soundcraft Vi console VM2 shows no list entries/devices at it‘s HiQnet list screen!

→ See FAQ 2-1

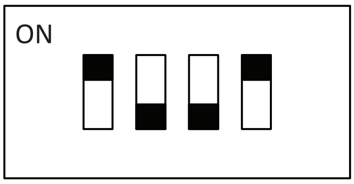
FAQ 2-4: What are the minimum requirements for a PC running System Architect?

- Windows 7, Windows Vista, Windows XP SP2, Windows 2000 SP4 or Windows 2003 Server SP1, Windows 7
- Only 32 bit Windows operating systems are supported
- Processor - 2 GHz (Dual Core)
- RAM - 2 GB
- Screen Resolution - 1024x768
- 200 MB Hard Drive space available

FAQ 3: My HUB 4000 Q has wrong network settings!

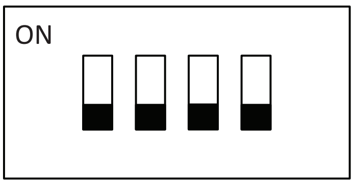
- Open Network Troubleshooter at System Architects – *Ribbon/Tools/Network/Network Troubleshooter* – and follow the instructions.
- If the Network Troubleshooter doesn't help open Readdress Devices at System Architects – *Ribbon/Tools/Network/Readdress Devices*
- The Readdress Devices panel is opened. There you can configure your devices network settings.
- If all of your HUB 4000 Qs are powered on and connected to the HiQnet network and no HUB 4000 Q is shown in the list at the Readdress Devices panel, your HUB 4000 Q has no network connection to System Architect (detection of devices needs up to 60 seconds).
- Please follow these steps:

    - On the right side of the HUB 4000 Q, behind the AKG logo, a DIP switch can be found which changes the IP address negotiation. Set the DIP switch to the following setting:



DIP 1, 4: ON - DIP 2, 3: OFF

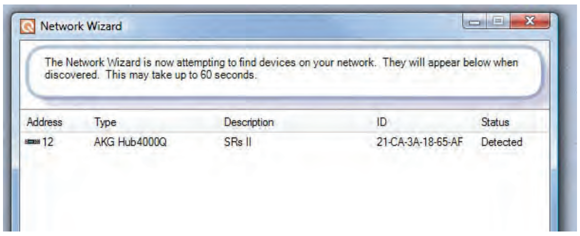
    - Power cycle the HUB 4000 Q (Switch the Power OFF and ON)
    - Wait till the 8 slot LEDs went from the left and right side to the middle and back again periodically.
    - Set the DIP switch to the following setting:



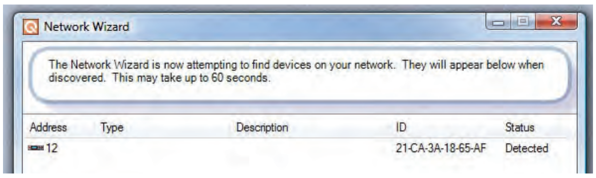
DIP 1, 2, 3, 4: OFF

    - Now the HUB 4000 Q starts first with trying to find an IP address with DHCP. If the HUB 4000 Q doesn't get an IP address over DHCP (if no DHCP server is connected to your HiQnet network) the HUB 4000 Q tries to get an IP address with AutoIP (IP range 169.254.1.1 to 169.254.254.255 with a subnet mask 255.255.0.0)
      → The data LED is blinking periodically every second.
      → All slot LEDs are off.
      → The address negotiation can take up to 5 minutes.

- Now the HUB 4000 Q should have a valid IP address
  → The data LED is showing the network traffic (it should blink non periodically and very fast)
  → The slot LEDs are ON if a AKG Device is connected
- If the HUB 4000 Q has a valid IP address, the Readdress Network panel should show up a new entry with your HUB 4000 Q because System Architect has now a network connection to the HUB 4000 Q
  - If the new entry has also entries at columns '*Type*' and a '*Description*' the network is configured right (see following picture).

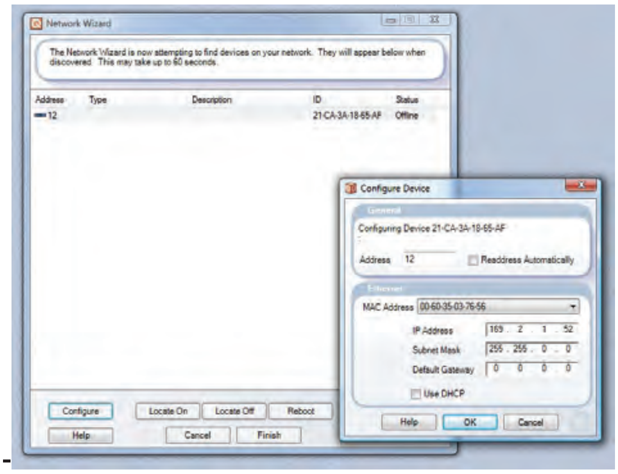


  - You can now start working with System Architect/AKG Wireless iPhone App, VM2, HUB 4000 Q and the AKG wireless system.
  - If the new entry has NO entries at columns '*Type*' and a '*Description*' the IP address and subnet settings of the HUB 4000 Q are wrong.



Please follow the next steps:

- Double click on the list entry or click the Configure Button at the bottom of the Readdress Devices panel:

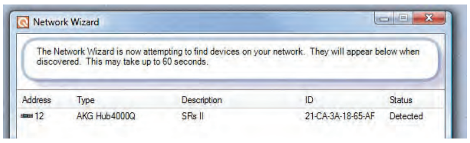


-

- The Configure Device Dialog is opened. At this example the Device has the IP address 169.2.1.52 and is in the subnet 255.255.0.0. The computer has in this example the IP address 192.168.1.1 and the subnet 255.255.255.0.

That means that the device has the wrong IP address and subnet. - See FAQ 6-1

- Now you can check the *Use DHCP checkbox* if you want to use a DHCP server for getting a valid IP address and subnet from the DHCP server
- Configure your PC operating system for retrieving automatically an IP address and subnet – See FAQ 14
- Or you type in a static IP address and subnet by hand, for example IP address 192.168.1.10 and subnet 255.255.255.0. See FAQ 6-2.
- Click OK and your HUB 4000 Q should have now the right IP address and subnet. The PC and the HUB 4000 Q should be now in the same subnet and should have valid and unique IP addresses
- Now the HUB 4000 Q should show up at Readdress Devices panel. The HUB 4000 Q is now configured right and should show up also at the HiQnet Explorer and can be used at the Venue View



Please also refer to System Architect Online help for further information on how to set-up a HiQnet™ network.

Chapter 'Troubleshooting' – Section 'Missing Devices'

FAQ 4: If my HUB 4000 Q is shown at Venue View, System Architect asks me to perform a firmware update of my HUB 4000 Q.

If you have downloaded a new version of System Architect and start System Architect and a HUB 4000 Q is detected you have to perform a firmware update.

Mixed configurations of an old HUB 4000 Q firmware together with the newest System Architect versoin are not supported. It is recommended to use always the latest version of System Architect which also includes the latest versions of HUB 4000 Q firmware.

Please download the latest version of System Architect from

http://hiqnet.harmanpro.com/downloads.php

FAQ 5: What is the IP address range of the HUB 4000 Qs AutoIP?

The IP range for the Auto IP function of the HUB 4000 Q is 169.254.1.1 to 169.254.254.255 with a subnet mask 255.255.0.0.

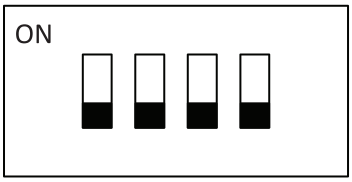
FAQ 6-1: What IP address is used by the HUB 4000 Q?

The IP address negotiation of the HUB 4000 Q depends on the DIP switch setting of the HUB 4000 Q. Also see FAQ 6-2, 6-3, 6-4, 6-5.

DIP switch settings:

On the right side of the HUB 4000 Q, behind the AKG logo, a DIP switch can be found which changes the IP address negotiation. DIP switch changes take only effect after the HUB is power cycled. It is recommended to use the default DIP switch setting (all switches OFF). This setting uses the stored IP address setting.

Configuration 1:

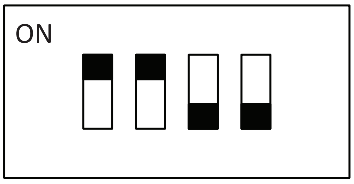


DIP 1, 2, 3, 4: OFF

Default, recommended configuration:

- Use stored settings – Uses System Architect settings
- At shipping – first start up: DHCP & AutoIP

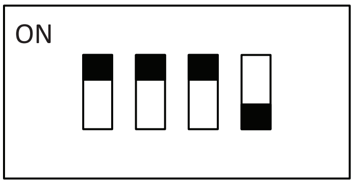
Configuration 2:



DIP 1, 2: ON - DIP 3, 4: OFF

- DHCP & AutoIP – Overrules System Architect setting

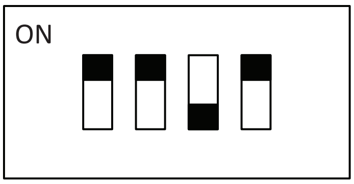
Configuration 3:



DIP 1, 2, 3: ON - DIP 4: OFF

- DHCP only – Overrules System Architect settings

Configuration 4:



DIP 1, 2, 4: ON -DIP 3: OFF

- AutoIP only – Overrules System Architect settings

FAQ 6-2: What IP address and subnet should i choose?

Normally the PC with System Architect, Wireless iPhone app or Vi console with VM2 are supposed to operate in the same HiQnet LAN (Local area network) as the AKG HUB 4000 Qs.

This means that the IP address of the PC, iPhone or Vi consoles must in the same subnet as the HUB 4000 Qs IP addresses. So for proper operation all subnets of all devices in the HiQnet LAN must be the same.

It is strongly recommended to use an Ethernet DHCP router in your HiQnet network. If you connect a DHCP network router (which has a built in DHCP server for automatic IP address management in your ethernet network) setting up the system is much more easier and faster because you don't have to care about IP addresses (that's done by the DHCP server inside your DHCP router).

Note: It is possible to operate several HUB 4000 Qs throughout several subnets from a single System Architect. This needs advanced network configuration and is only recommended for advanced users. Please contact HiQnet@akg.com for further assistance.

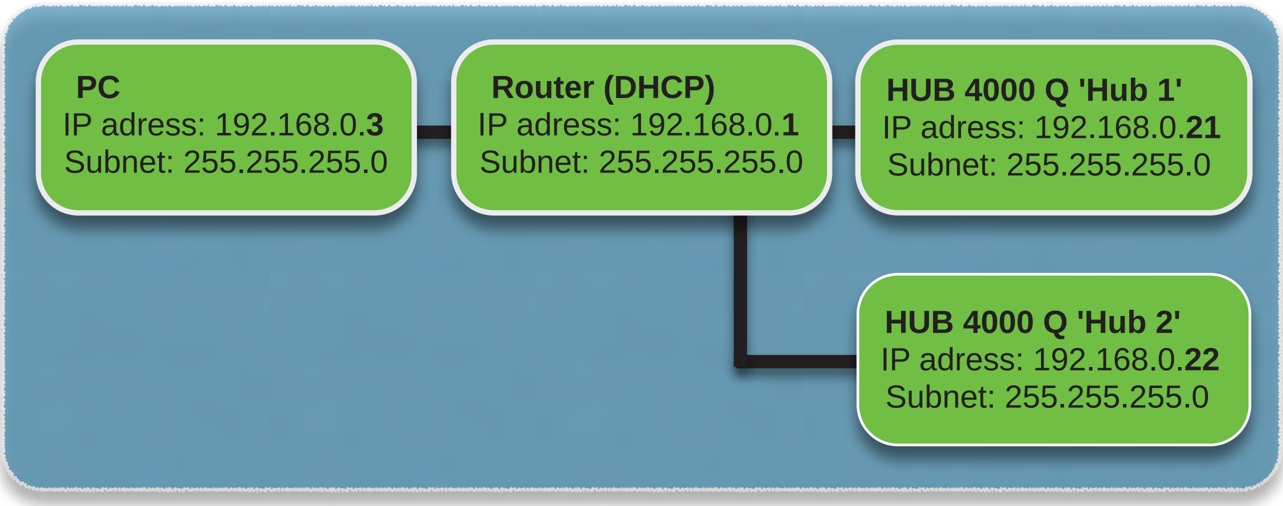
IP address: Each device in a LAN must have a unique IP address

Subnet: The subnet defines the digits of the IP address which are unique in a subnet

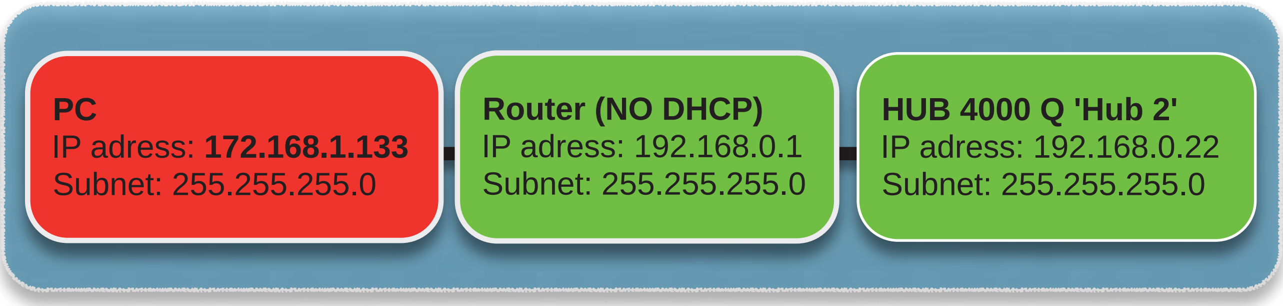
EXAMPLE: If the subnet is 255.255.255.0 The first 9 digits of all IP addresses in the subnet have to have the same numbers. They define the subnet. Only the last 3 digits of the IP addresses are allowed to be different.

Devices with IP addresses like 192.168.0.1, 192.168.0.123, 192.168.0.204 would be in the same subnet (The subnet is then 192.168.0.xxx) whereas a device with a IP address of 172.168.0.3 won't be in the same subnet as the first digits are not 192.168.0.xxx

EXAMPLE: The following example shows a HiQnet system with a PC running System Architect, a DHCP router for automatic IP address management and 2 HUB 4000 Qs. All IP addresses and subnets are configured right. System Architect will discover the HUB 4000 Qs instantly and you are ready to work with the AKG wireless system.



EXAMPLE: The following example shows a HiQnet system with a PC running System Architect, a router (DHCP turned off) and one HUB 4000 Q. The PC is in the right subnet but it's IP address is configured wrong. System Architect will not discover the HUB 4000 Q because there is no proper network connection between the PC and the HUB 4000 Q. The IP address of the PC should be 192.168.0.10. Then the PC would be configured right.



FAQ 6-3: Do i need to configure my Windows firewall?

It is recommended to switch of your Windows firewall.

FAQ 6-4: How do i setup my network router?

Normally you can do this within your internet browser by typing in the IP address of your wireless router in the adress bar of internet explorer. By default most router use the IP address 192.168.0.1.

Please refer to the documentation/manual of your network router.

FAQ 6-5: How should i configure my network?

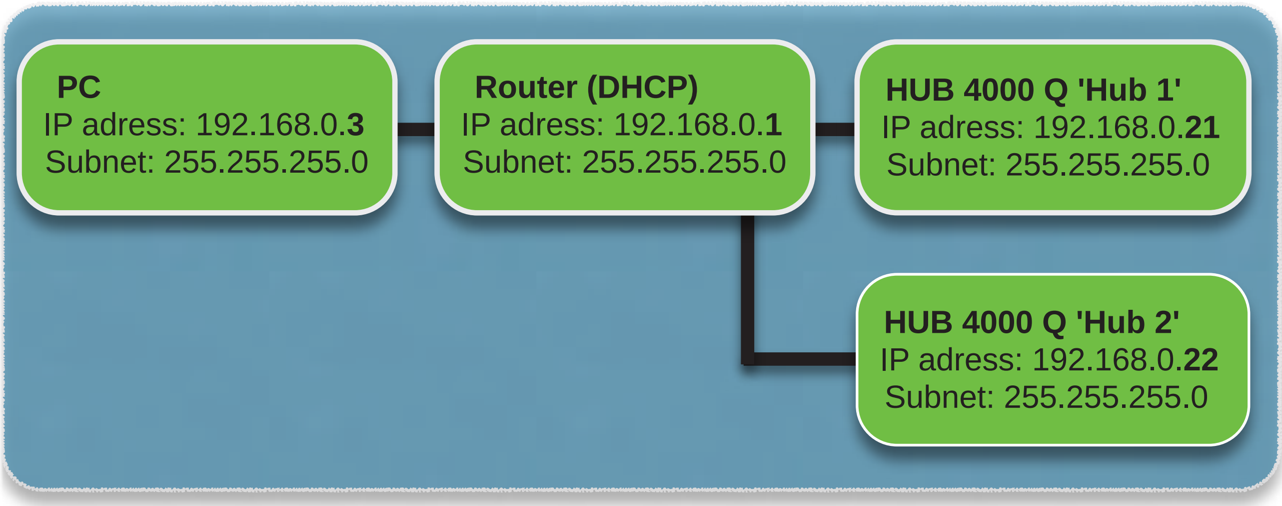
There are multiple possible configurations which are possible. The three most common configurations are:

1 - PC, multiple HUB 4000 Qs, DHCP router

This is the recommended configuration. The PC is connected via a network router with DHCP to the multiple HUB 4000 Qs. The network router takes care of the IP address and subnet settings of the PC and HUB 4000 Qs.

Make sure that the PC and the HUB 4000 Qs are configured to receive IP settings via DHCP.

See FAQ 3, 5, 6-1, 6-2, 6-4, 14.

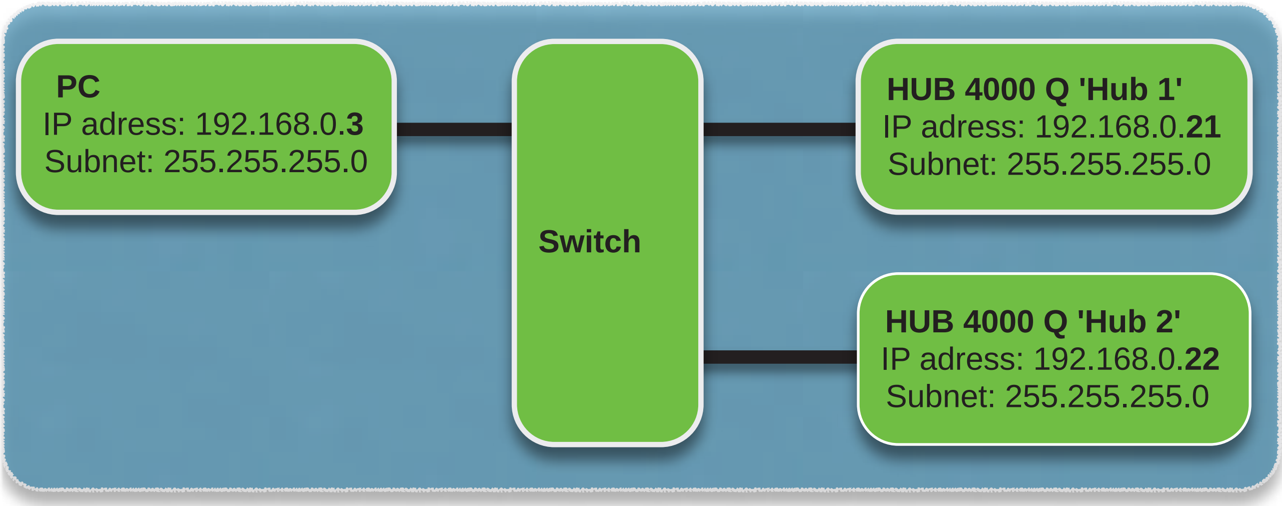


2 - PC, multiple HUB 4000 Qs, network switch

The PC is connected via a network switch to the multiple HUB 4000 Qs. The network settings must be set manually at the PC and HUB 4000 Qs.

Make sure that the PC and the HUB 4000 Qs have the right IP settings.

See FAQ 3, 5, 6-1, 6-2, 6-4, 14.

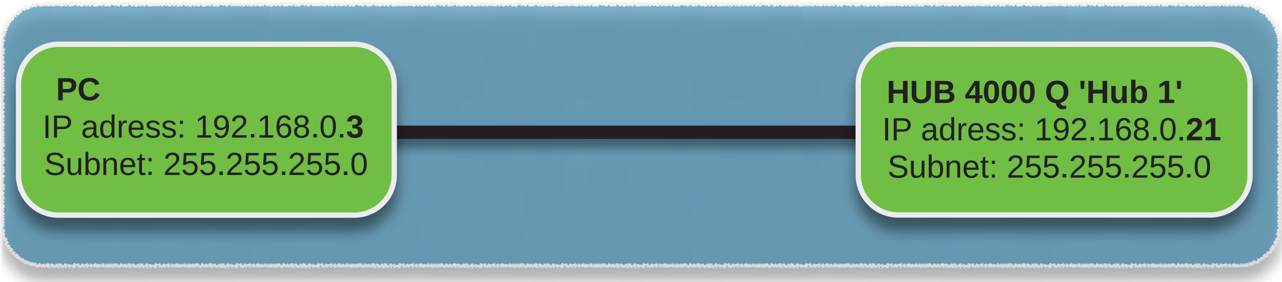


3 - PC, a single HUB 4000 Qs, connected via a crossover network cable

The PC is connected via a crossover network cable to a single HUB 4000 Q. The network settings must be set manually at the PC and the HUB 4000 Q.

Make sure that the PC and the HUB 4000 Qs have the right IP settings.

See FAQ 3, 5, 6-1, 6-2, 6-4, 14.



FAQ 7: How long does it need till the HUB 4000 Q gets an IP address

Be aware that the complete IP address negotiation process can last up to 5 minutes.

The first time the HUB 4000 Q receives an IP address, the address is stored in the device's memory. The next time the HUB 4000 Q starts, the HUB 4000 Q uses the stored IP address if DIP switch configuration 1 was choosen (see FAQ 6-1). That saves a lot of start-up time.

FAQ 8: What is the functionality of the eight slot LEDs:

Each of the eight slots of the HUB 4000 Q has a dedicated blue slot LED on the front of the HUB 4000 Q. The slot LED is off if no device is connected to that slot. The slot LED lights nearly permanently (actually corresponding to the net-traffic) if a device is connected to the HUB 4000 Q and the device is turned on. If the device is turned off the slot LED blinks periodically to indicate that the device is turned off.

FAQ 9: I see 2 HUB 4000 Q at the Venue View, but if I click on one HUB 4000 Q icon I just can control one HUB 4000 Q.

To be able to control multiple HUB 4000 Qs from a single window you have to create a Master Control Panel. Read through FAQ 10 about how to create a Master Control Panel.

FAQ 10: How can I control multiple HUB 4000 Qs within a single panel?

CREATE a MASTER CONTOL PANEL!
Select all HUB 4000 Q icons at Venue View which should be added to the Master Control Panel. Right click on one of the selected HUB 4000 Q icons. Select from the Context Menu 'Create Master Control Panel' – 'AKG HUB 4000 Q'. Then a Master Control Panel for all the selected HUB 4000 Qs is created. The Master Control Panel is able to control all AKG devices of all HUB 4000 Qs from a single panel.

FAQ 11: After System Architect is coming online a stripe with red background is showing up on the HUB 4000 Q product panel

The red background of a Stripe indicates a MISMATCH between the physical device, connected to the HUB 4000 Q and its dedicated Stripe.

The Mismatch can be resolved in two ways:

1. Resolve at System Architect:

Double click on the Mismatching stripe at the 'Resolve Mismatch' button. The Mismatch will resolve by deleting the offline plug-in Stripe and retrieving all information of the physical AKG Device connected to the hardware HUB 4000 Q and creating a new matching Stripe for that physical AKG Device.

ATTENTION: All settings of the offline Stripe will be LOST!

2. Resolve at the physical HUB 4000 Q:

User can resolve the Mismatch by changing the physical AKG Device which causes a Mismatch.

The 'Expected device' label gives you information about the expected device which was configured offline. Take a note which physical AKG Device is expected with which Band Variant, at which Slot. Disconnect the Mismatching physical AKG Device from the hardware HUB 4000 Q and connect a physical AKG Device of the Device Type with the Band Variant of the expected physical AKG Device. The Mismatch at the plug-in will be automatically resolved.

FAQ 12: The HUB 4000 Q is going Online/Offline after some time – or – The Meters, buttons and other controls are not/or very slow reacting

See the FAQ 13, 2-4!

FAQ 13: The HUB 4000 Q is part of a Cobranet System and behaves very strange. The HUB goes Online/Offline, is not or slow reacting

If the HUB 4000 Q is part of a Cobranet System the network settings must be set very carefully. If Cobranet Broadcast messages/streams are used the HUB is spamed with these messages and cannot receive or send HiQnet messages fast enough.

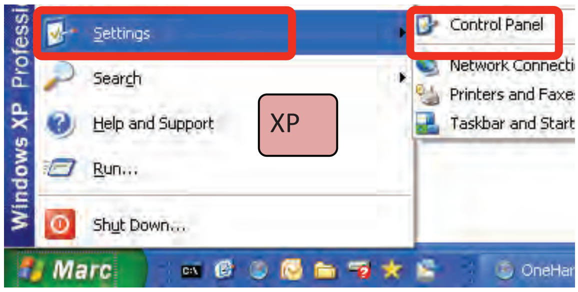
It is recommended to setup a separated VLAN for all HiQnet devices like the HUB 4000 Q and a separate Cobranet VLAN. Then the Cobranet messages are not received from the HUB 4000 Q.

You can also use networkswitches which support Ethertype filtering and filter the Cobranet Ethernet messages at the port of the switch to which the HUB 4000 Q is connected.

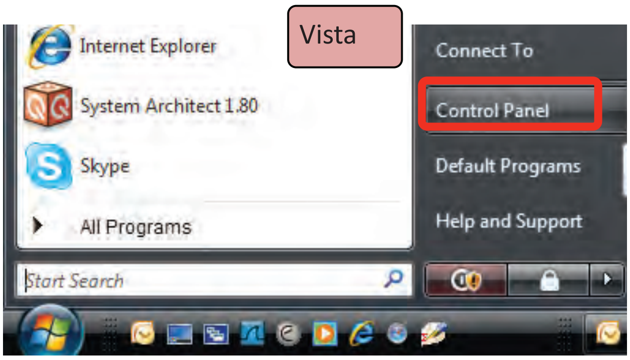
If you need further help on this issue please contact: hiqnet@akg.com

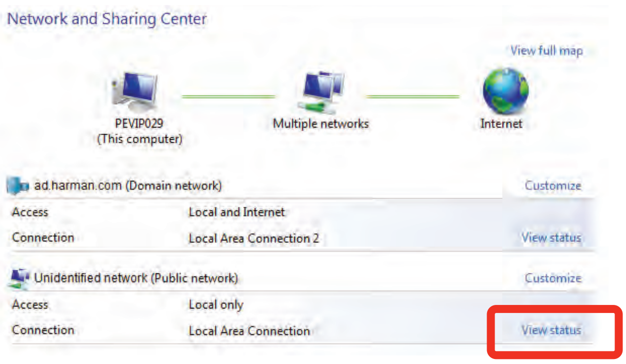
FAQ 14: How can I set the IP settings of my PC?

- The IP settings of your PC can be found at the Windows Control Panel/Network Settings.



  - Windows XP:
    - Click *Start Button/Settings/Control Panel*
    - At *Control Panel* click at *Network Settings*
    - At *Network Settings* double click the HiQnet network
  - Windows Vista:
    - Click *Start Button/Control Panel*

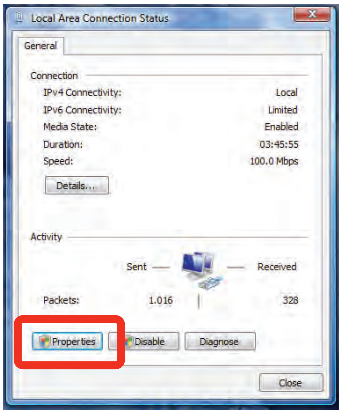




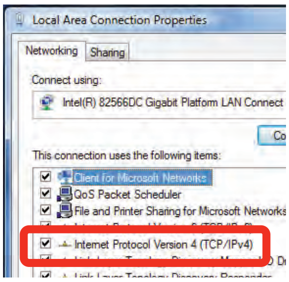
    - At *Control Panel* click on '*Network and Sharing Center*' and click on the '*View status*' link of the HiQnet network (*Unidentified network* at this example)

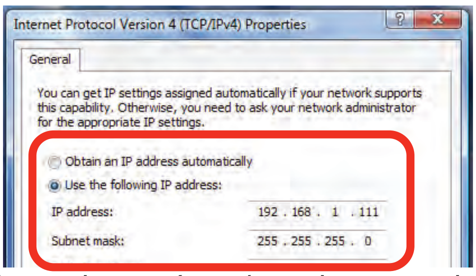
- Windows 7:
  - Click *Start Button/Control Panel*
  - At *Control Panel* click on ‘*Network and Sharing Center*’ and click on ‘*Connections*’ link of the HiQnet network

- This opens the *Local Area Connection Status* panel. Here you can see the status of your network.



- Click at the *Properties* Button
- Double Click at the *Internet Protocol 4 (TCP/IPv4)* entry





- At the *Internet Protocol Version 4 (TCP/IPv4) Properties* panel you can set the PCs network settings for this network.
  If you choose ‘*Obtain an IP address automatically*’ the PC tries to get an IP

address with DHCP or AutoIP.
You can also define a static IP address and subnet at '*Use the following IP address*'. See FAQ 6-2 for more information about setting an IP address

FAQ 15: System Architect crashes when starting 1 Click Setup!

System Architect 1 Click Setup only works with 32 bit versions of Windows operating systems. With 64 bit versions of Windows operating system System Architect is crashing when launching 1 Click Setup. This issues is known and AKG is working on a solution to fix it.

FAQ 16: Environment Scan doesn't work, or is very slow!

Normally your PC doesn't fullfil the minumum requirements if environment scan doesn't work. Please check your PCs specification and the minumum requirements at FAQ 13+2-4.

FAQ 16-2: At devicegrid at column 'Band' 'RF Error' is shown! At Stripe Info Menu ,'RF Error' is shown!

This means that the band is not supported by System Architect. Please contact HiQnet@akg.com for further assistance.

FAQ 17: SST 4 is not able to scan!

The SST 4 is a stereo stationary transmitter of an In Ear Monitoring system which means that it is only able to transmit RF signals. It cannot receive RF signals and for that reason it is not able to perform an environment scan. If you want to perform an environment scan at the bands of your SST 4s you need to connect either SR 4000/4500 or DSR 700 to your HUB 4000 Qs.

FAQ 18: I cannot update the firmware of SR 4000!

The firmware of SR 4000 cannot be updated due to technical reasons.

FAQ 19: After loading a venue file audio and RF meters are not working any more!

This is a known issue. AKG will fix this bug as soon as possible. However closing all Custom control panels and docking and floating the product panel or master control panel solves the problem.