



BROADWAY Contents



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BROADWAY

1 Introduction

1.1 **BROADWAY Introduction**

Live Sound

There is an awkward paradox at the heart of live sound mixing today.

On the one hand, there is a growing requirement in live sound for reset automation and repeatability in the increasingly technical environment of the Live Event. Not only does the sound designer need to be able to reset and re-route channels reliably during the evening to get the most out of the available resources, but there is now a need for integrated control over external devices such as FX units, Playback and outboard processing. Engineers and operators are becoming sidetracked from mixing duties by the need to set up, reset, trigger and check all these extra parameters, both inside the console, and externally.

All live engineers know that immediacy of control and ease of use in a high-pressure situation are of paramount importance for any piece of live sound equipment. There is no point in developing a highly complex automation system if the engineer can no longer creatively mix the show without worrying about complex operational tasks and special software requirements.

Secondly, the number of radio microphones in use on even quite basic shows is constantly increasing, and conventional console frames are having to be extended to allow for more and more input channels.

To further complicate matters, the Producer or venue owner will not want to give up any more seats than is absolutely necessary.

Broadway is our answer to these changing needs. Whilst offering the designer a high level of reset, programmability, outboard control and flexibility, Broadway remains very simple for the engineer or operator to use on a daily basis, and furthermore can be configured to take up much less space than a conventional console of an equivalent number of inputs.

Once Broadway has been set up for even the most demanding live show, the main operational areas of the console - the fader trays, the channel strip and the metering - are sufficiently familiar to allow the operator to get on with the task of creatively mixing the show, while the mundane daily switching and routing functions are performed by the console on a scene-by scene basis.

System Overview

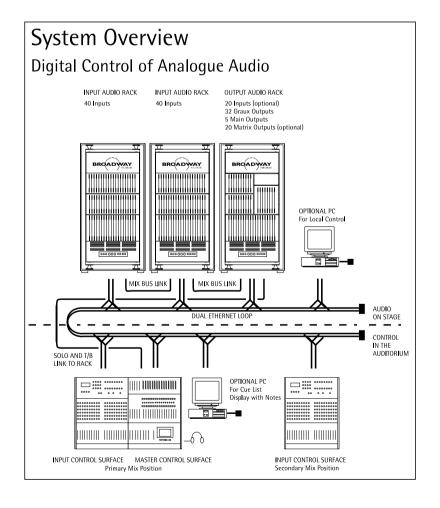
The Broadway system is based upon an "X into 32 + 5 into X" configuration. This means that the console will support potentially any number of inputs, mixed onto 32 Group / Aux (GrAux) send busses, from which a matrix of up to 40 outputs may then be derived.

There are five basic building blocks for any Broadway system:

- Input Audio Rack
- Master Audio Rack
- Input Control Surface
- Master Control Surface
- VCA Extender Surface

The above units may be arranged in any order or physical location, but must be interconnected via the Ethernet Network. Every element, or "Node", on the Network is a "stand-alone" unit. Each has its own Power Supply (dual redundant in the racks), and a Processor Card for data processing and Network connectivity.

Please see "Running The System" below for a more detailed description of Network Functionality and system boot procedure.



BROADWAY Introduction 1.3

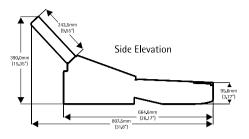
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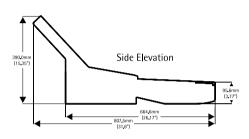
2.1 **BROADWAY Installation**

Dimensions and Configurations

Input Control Surface



Master Control Surface



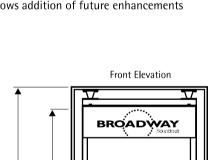
600mm (23.62") 12.0mm (0.27")

600mm (23.62") 12.0mm (0.27")

Input and Master racks share the same dimensions

Rack shown mounted in suspended flightcase

22U Frame allows addition of future enhancements

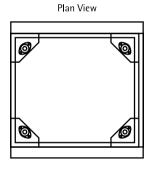


1175mm (46.26")

978mm (22U)

482.6mm (19")

575mm (22,64")



Side Elevation

The state of th

2.2

Precautions and Safety Instructions

General Precautions

Avoid storing or using the system in conditions of excessive heat or cold, or in positions where it is likely to be subject to vibration, dust or moisture. Do not use any liquids to clean the fascia of the unit: a soft dry brush is ideal. Use only water or ethyl alcohol to clean the trim and scribble strips. Other solvents may cause damage to paint or plastic parts.

Avoid using the console close to strong sources of electromagnetic radiation (e.g. video monitors, highpower electric cabling): this may cause degradation of the audio quality due to induced voltages in connecting leads and chassis.

Caution!

In all cases, refer servicing to qualified personnel.

Handling and Transport

Be sure to disconnect all cabling before moving. At all times avoid applying excessive force to any knobs, switches or connectors.

Power Supplies

The power supplies in the racks are auto adjusting for mains input voltages. The power supplies in the control surfaces are user switchable between 2 input voltages: nominally 115V AC and 230V AC.

In all cases ensure that the correct fuses are used. The values required are clearly shown on the appropriate fascias.

Signal Levels

It is important to supply the correct input levels to the racks, otherwise signal to noise ratio or distortion performance may be degraded; and in extreme cases, damage to the internal circuitry may result. Likewise, on all balanced inputs avoid sources with large commonmode DC, AC or RF voltages, as these will reduce the available signal range on the inputs. Note that 0dBu = 0.775V RMS.

The microphone inputs are designed for use with balanced low impedance (150 or 200 ohms) microphones.

Caution!

DO NOT use unbalanced microphones or battery powered condenser microphones without isolating the +48V phantom power: degraded performance or damage to the microphone may result.

The sensitivity of the Mic inputs is variable from +4dBu to -66dBu, with a maximum input level of +25dBu. The Line Input sensitivity is variable from +14dBu to -46dBu with a maximum input level of +35dBu.

BROADWAY Installation 2.3

Mains Installation

Wiring Considerations

- A For optimum performance it is essential for the earthing system to be clean and noise free, as all signals are referenced to this earth. A central point should be decided on for the main earth point, and all earths should be "star-fed" from this point. It is recommended that an individual earth wire be run from each electrical outlet, back to the system star point to provide a safety earth reference for each piece of equipment.
- **B** Install separate mains outlets for the audio equipment, and feed these independently from any other equipment.
- **C** Avoid locating mains distribution boxes near audio equipment, especially tape recorders, which are very sensitive to electromagnetic fields.
- **D** Where possible ensure that all audio cable screens and signal earths are connected to ground only at their source.

Power Supplies

Warning!

Before switching on your system, check that the mains voltage selectors on the control surfaces are set to the correct mains voltage for your area, and that the fuse is of the correct rating and type. This is clearly marked on the case of the power supply. Do not replace the fuse with any other type, as this could become a safety hazard and will void the warranty.

2.4 BROADWAY Installation

Connections

Soundcraft part numbers are in italics

Master Audio PFL

Console has Neutrik NC5MP 5 pin male XLR connector (FK0933) which mates with Neutrik NC5FX (FK0934)

Pin Signal pin I Audio Gnd pin 2 Left channel + pin 3 Left channel pin 4 Right channel + pin 5 Right channel -

Master Audio - Rack Audio Multicore

Console & Rack have ITT Cannon 23 pin Trident Ringlock connector part no.192990-1290 (FF0934).

Mating part: ITT part # 192990-1320 (FF0933).

Need male pins 192990-0080 (FG0664) for both ends of audio cable

Pin	Signal		
I	PFLIL+		
2	PFLIL-		
3	CONN		
4	PFLIR+		
5	PFLIR-		
6	AGND		
7	PFL2L+		
8	PFL2L-		
9	PFL2R+		
10	PFL2R-		
11	PFL3L+		
12	PFL3L-		
13	PFL3R+		
14	PFL3R-		
15	CRML+		
16	CRML-		
17	CRMC+		
18	CRMC-		
19	CRMR+		
20	CRMR-		
21	AGND		
22	TBOUT+		
23	TBOUT-		

Note: pin 3 tells the console that the rack is connected.

We recommend Canford Audio FSM12 cable.

BROADWAY Installation 2.5

Littlite

12V 300mA Halogen type.

Console has 4 pin Neutrik XLR female connector NC4FX

pin 4 + 12V dc (300mA)

pin 2& 3 not connected

pin I 0V (chassis)

Littlite must be at least 18" to reach over ACS.

We recommend Littlite part # 18RXH4

ACS

25 pin D-type connector

Wired pin to pin. All pins must be connected - i.e. standard "printer" cables cannot

be used.

Male part: FB0170 25 pin D-type IDC cable mount

Female part: FB0101

Recommended cable: Thomas & Betts part # 300-25-100

N.B.: maximum cable length we have tried is 10m.

Ethernet

Broadway uses 50W BNC (10-base 2) connections (to IEE 802.3)

Recommended cable: RG58BU type (UR43) (LB0119)

Minimum cable length: 3m Maximum cable length: 100m

Recommended connector: M/A Com part # 115-01-010 (FZ2260)

50W Terminator @ end nodes only: M/A Com part # B35Z98E501X99 (FZ2259)

"T"-pieces - M/A Com part # 105-03-000 (FZ143768)

Audio Inputs

The audio inputs are via combined XLR/I/4" Jack Neutrik connectors (NCJ6FK).

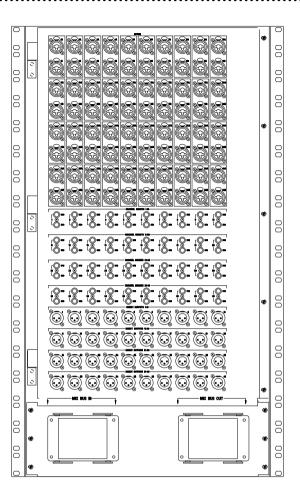
2.6 BROADWAY Installation

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3 Components of the **System**

Input Audio Rack

Hardware



An Input Audio Rack contains 40 complete inputs, along with all the connectivity both for the Audio and for the Network. Potentially any number of Input racks may be added to a Broadway system, each rack adding 40 inputs. The MIX BUS interconnections are carried via HAN144D 144-pin multiway connectors from rack to rack, ending up in the Output rack. These connectors must be attached (using the supplied interconnect cables) to enable audio to pass through the system.

Audio connectivity in the standard console is via Neutrik Combi connectors, which offer both female XLR and 1/4" TRS Jack sockets. Certain custom consoles will include HAN144D or EDAC90 multiway connectors in addition to the standard connectivity.

The inserts are on 1/4" TRS Jacks, are fully balanced, and switchable IN/OUT via software from the surface.

The electronics inside the rack consist of vertically-mounted PCB boards, and are divided into three sections - input cards at the top, EQ cards in the middle, and Mix cards at the bottom. The racks seat twenty cards from left to right, and each card in the input and EQ sections carries two channels - hence 40 channels per rack.

In the lower part of the rack is a sliding tray which contains the dual redundant power supplies and the processor card. The tray, when removed from the rack, will break the contact with the rest of the rack, and therefore isolate the supply to the rack. (I) The Processor card is attached to the front of the rack, and will therefore be disconnected with the movement of the tray.

The power supplies operate in parallel, but the rack can operate on a single supply if one should fail. The processor card contains all the network connectivity, and the MIDI IN / OUT and THRU.

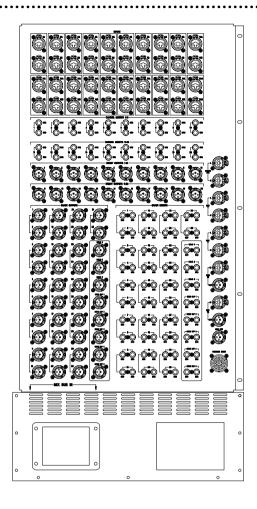
The input racks house 40 inputs each (or 20 inputs if half-filled). The front panels of the rack are sectionalised and hinged, and may be opened to reveal the Input, EQ and MIX cards respectively.

A series of led's on the cards show the status of the data and power loops. Green LED's represent the power, and red the data loop. All power, audio and data loops are supplied via the backplane connection.

¹ Of course, as with any Broadway component, the unit should be isolated from the mains before attempting to remove or modify any hardware element of the unit.

Output Audio Rack

Hardware



The output rack carries all of the GrAux, Main and Matrix outputs for the Broadway system. It also handles connectivity to the Solo, Talkback and comms audio on the Master Surface via a 23-pin connector.

The electronics are on vertically-mounted PCB boards, similar to the Input racks. The cards are divided into two main types - Mix Bus and Quint Output. The rearcon of the output rack contains all output connectivity including inserts for the GrAuxes and Matrices. Note that the GrAux inserts may be switched IN/OUT in software, but the Matrix inserts are broken upon insertion of a jack, and should therefore be run through a normalised patchbay or processor to maintain the signal.

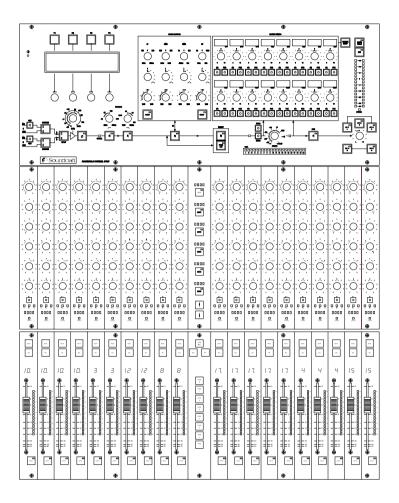
The power supply and processor are the same as those in the input rack. See the description in the input audio rack section above.

The output rack handles all bus outputs from, and data routing for, the Broadway system. Because of this, it is the only potentially vulnerable part of the system, and it is recommended than an Uninterruptible Power Supply (UPS) be installed on this unit. As on the Input rack, the front doors of the rack are sectionalised and hinged, and may be opened to reveal the MIX and QUINT OUTPUT cards inside.

A series of led's on the cards show the status of the data and power loops. Green LED's represent the power, and red the data loop. All power, audio and data loops are supplied via the backplane connection.

Input Control Surface

Hardware



The input control surfaces house 20 motorised faders, each with associated Mute, Solo and SEL(ect) switches, and 4-character dot matrix name displays. There are 8 "BANKS" of faders available on each Input surface, selectable from the Bank switches in the centre of the fader tray.

Above these, in the Encoder Tray, are some 120 encoders (6 per channel), which are assignable to perform any rotary channel strip function.

At the top of the Input Surface is the Assignable Channel Strip (ACS). This offers a single complete input channel strip, and will display and edit the parameters of the SEL(ect)ed channel. The ACS is removable, and may be carried up to approximately 100 metres away from the console for more effective console setup, allowing the user to listen in different areas of an auditorium whilst making changes.

The rear panel of the input surface offers IEC mains input, the main power switch, voltage selection, fader and main power fuses, serial and barcode number labels, and the local connector to the ACS.

The Input Surface also has two removable "trays" in the rear panel: a Power Supply and a Processor Card. These trays may be removed by the user when servicing or upgrading the surface. The power supply tray, once mated with the power distribution backplane inside the Surface, organises the routing of appropriate power to the relevant areas of the unit.

The processor tray contains the main processing power of the surface, and contains all the interconnection ports required for network and system connection: C3, ESBUS, MIDI IN, OUT, THRU, and BNC network connections.

Functional Description - Modes

Input Control Surfaces operate in one of four modes. These modes dictate the functions performed by the hardware and software, and are selected via the three Mode switches in the centre of the Input surface.

Though the Input surface is normally used to control the levels, EQ, and routing of any twenty input channels, it may alternatively be used as the control surface for GrAux (Group / Aux) or Matrix outputs, enabling inputs, effects sends, foldback feeds and group to matrix mixes to be set up from a single control surface. A fourth mode enables the 20 faders on the Input surface to be used as 20 VCA master faders, giving the user access to all 20 VCA groups on a single surface.

Input Mode

The most basic of the four modes is Input mode. In this mode, the surface offers complete control over up to 8 Banks of 20 inputs (i.e. 160 channels).

ACS

The Assignable Channel Strip gives the operator a view of every function of a particular stereo or mono input channel, simply by pressing the SEL (ect) button on that channel. Dedicated controls are offered for every function, with the GrAux routing controlled by sixteen groups of controls which may be selected to represent GrAux I-16 or 17-32. Each GrAux is identified by name by the four-digit displays above each control. The four-band fully parametric EQ is accessed by dedicated rotary controls and displays; this may be displayed in graphical form on the ACS LCD screen. This display is also used to identify the channel being affected by the ACS, and may be used in conjunction with the four multi-function rotary controls and switches for a variety of further facilities.

Panning is provided between any combination of the five main busses, enabling true LCR, LCRS and quad images to be created. Dual meters are provided for use where channels are linked in stereo.

The ACS is normally located at the top of each Input Control Surface, but may be demounted and used remotely, connected to the main surface via a simple multiway cable. Since the currently selected channel may be changed on the ACS itself, and the majority of functions accessed from this small panel, this facility enables the user to "take the console into the auditorium", and set up level sends, EQ etc. whilst seated where it really matters - i.e. where the audience will be.

The channel under the control of the ACS may be isolated from the rest of the console so that it will not respond to changes caused by recalling new scenes, and may be "locked" to prevent another operator from inadvertently selecting a different channel to act under ACS control.

Encoder Panel

This section of the Input Control Surface contains six assignable rotary controls for each of the channels displayed. Indication of EQ, Inserted Dynamics (2) and Stereo status are provided, as well as a four-digit display showing the name or number of each channel. Each row of controls acts on a single function of the twenty inputs, such as GrAux sends, EQ parameters, or input gain: the function of the rows may be "scrolled" using "i" and "Å". The eight Control Assignment buttons located on the Master section are 8 user-programmable memory locations, into which the operator may store 8 favourite arrangements of the Input Surface Encoder Tray (i.e. 8 "Views" of chosen functions).

The function of a particular row may be "locked" to prevent its deselection, and any channel may be isolated to protect it from scene changes.

Fader Panel

The fader panel is the last of the three component sections of the input control surface. A SOLO button is located below each of the faders. This button operates much as the Solo on a conventional desk. The primary metering for each channel is adjacent to the 100mm motorised fader.

The twin seven-segment LED's above each fader display the VCA to which the channel is currently assigned, or, if multiple VCA's have been assigned, the Primary VCA (i.e. the lowest-numbered) is shown, and a decimal point appears after the second numeric.

The SEL(ect) button is used to bring the relevant channel up onto the ACS. This button is also used for various channel interrogation and configuration functions. These modes are selected from the touchscreen in the Master Section. The MUTE button operates as on a conventional console, muting all sends from that channel, although there is a facility for each GrAux which allows the user to set that buss to Pre-Mute when it's Pre-Fade. If the GrAux is set as Pre-Mute, the MUTE switch on the Input strip will have no effect on the pre-fade GrAux sends from that channel. This is the default setting.

GrAux Mode

Normally used in "Input" mode, if the Input Control Surface is selected to "GrAux mode", the 20 faders become the first 20 (or, on bank two, the last I2) GrAux master faders, and the six rows of encoders above each fader become the sends from that GrAux INTO the appropriate matrices.

ACS

In GrAux mode, the ACS will control routing of the currently selected GrAux to any of the Matrix outputs. The 16 "send" encoders will now carry the names of the Matrices, and will allow send levels to the first 16 matrices to be set, and the first touch of [SWAP] will bring up Matrices 17-32. A third, fourth etc. touch of SWAP will bring up 33-X, where X is the number of Matrix outputs. Insert In/Out, Fader level, Mute, Metering and Solo will all function as before, but all other functions will be disabled. The LCD may be used to change the currently-selected GrAux. The 5 main output switches and the PAN function will act as before, and will allow GrAuxes to be routed to the MAIN OUTPUTS.

² Integrated dynamic processing was not available at the time of writing, but all surfaces have been designed to cope with the arrival of this functionality should the need arise, either in the Broadway itself, or via remote control (see AppendixA).

Matrix Mode

Conversely, in Matrix mode, the 20 faders become the first 20 matrix masters, and the rotaries display and edit the contributions FROM the appropriate GrAuxes, scrollable through all GrAuxes as required.

ACS

In Matrix mode, the ACS will set feeds to the currently selected Matrix from any of the GrAux outputs. The 16 "draw" encoders will allow levels from the first 16 GrAuxes to be set, and the first touch of [SWAP] will bring up GrAuxes 17-32. Fader level, Mute, Metering and Solo will all function as before, but all other functions will be disabled. The LCD may be used to change the currently-selected Matrix.

VCA Mode

VCA mode puts the 20 VCA groups onto the 20 input surface faders, with the first 4 non-zero characters of the VCA group name appearing above the appropriate faders

When the input surface is in VCA mode, the ACS may still be used to select and edit input channels, as long as CHAN HOLD is selected on the ACS.

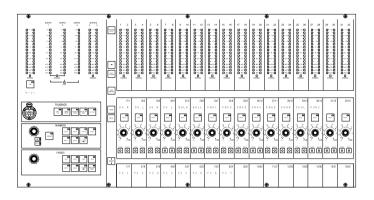
Why Bother With Modes?

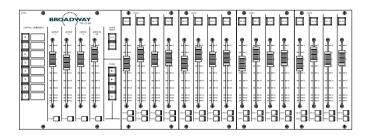
These modes, combined with the assignability of the console, allow the full audio path of the largest possible system (100+ into 32+5 into 40+) from input to matrix output, to be controlled from a single 600×800 mm footprint Input Control Surface. This is the extreme, of course, and most users will doubtless choose to have more Input Surfaces to have greater visibility at any given time, but the flexibility to have all or just 20 channels visible depending on space, financial resources etc., allows the designer and producer of a show to work together to provide a desirable but affordable audio system.

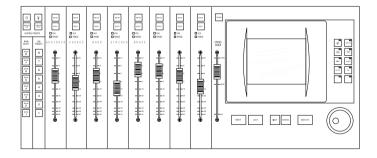
It might be appropriate to have, say, 60 physical faders during the production period, and just 20 when the show is up and running, thus reducing running costs once the design has settled down. The spare Input Surfaces could then be placed back into hire stock, and sent out with a different Master Surface for the next event.

Master Control Surface

Hardware







The Master Surface provides all the necessary hardware for control over the 32 GrAux outputs, the 5 Main outputs and the Matrix outputs (if fitted).

The fader tray contains 8 VCA Master Faders, with 8-character dot matrix name displays and associated SOLO and MUTE switches. There are also 8 BANK master switches and 8 Preset switches.

To the right of the VCA faders lies the touchscreen and associated page and cue control switches. This area is dedicated to the sceneset automation system, and is described in more detail in the Sceneset Automation section below.

Above the fader tray is the GrAux Fader Panel, which contains 16 Faders for the GrAuxes and 4 Faders for the five Main Outputs (this will be explained later).

The top section of the Master Surface houses the Metering for the 32 GrAux outputs and the 5 Main outputs. It also carries the talkback and communications switches.

Meterbridge

This allows metering of all 32 GrAuxes, plus the main outputs. Via a MTX METER switch, it is also possible to swap the GRAUX meter section to display 32 matrix outputs. The STEREO led below each pair of matrices denotes that the pair of GRAUX or MATRIX outputs is linked into stereo.

MASTER MUTE sets all MUTE switches on the system to MUTE ON. This function is disabled in LIVE mode.

LIVE locks out those functions which could be detrimental to a live performance, such as MASTER MUTE, talkback, solo in place and cue editing.

GrAux Fader Panel

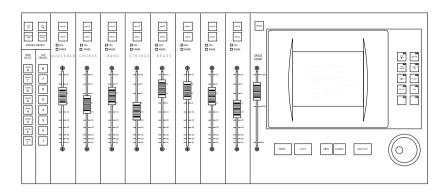
These controls allow access to the 32 GRAUX or MATRIX outputs (switchable GRAUX/MATRIX in banks of 16). The first 16 outputs are controlled via faders, and the last 16 are available on rotary encoders. Each output has a 4-character LED matrix display for name information, a MUTE button, a FUNC (tion) button, and a SOLO button. SOLO operates as on the input faders. The FUNC button toggles the current function on or off - the function mode being selected via the four FUNC buttons to the left of the 16 GRAUX master faders. When in INS mode, for example, the FUNC button toggles the insert IN/OUT for its associated output. For a full description of the FUNC switch, please see 4.5.2 below.

Talkback and Monitoring functions are available from this panel. All master outputs, two external inputs plus the listen mic may be monitored, with a self-cancelling or additive law, depending upon function. Talkback may be performed with an External system, a monitor desk, the audio racks and the comms loop. The user may set up talkback to desired GrAuxes and Matrices, via the use of the FUNC button in TB ASSN mode. TALK TO BUSS then arms all the relevant outputs for talkback.

In addition to the 16 GRAUX faders, there are four Master Output faders, controlling outputs 1, 2, 3, and 4/5. The last fader is paired to facilitate stereo surround level setting.

The 8 control assignment buttons allow the user to store 8 user-defined arrangements of the 6 assignable encoders on the input surfaces.

VCA Fader Panel



This houses the 8 VCA faders. Of course, being a digital control system, the term "VCA" on the control surface is not strictly correct - these are master faders for "links" or "groups" of other faders, and the value of the master fader will be added to all slave faders. Each input or GrAux fader may be assigned to one or more of the 20 VCA groups per cue. The seven-segment display above each of the input faders will show the "primary" VCA to which that input is assigned - that is, the lowest numerical value VCA. If the input is assigned to more than one VCA, a small dot is illuminated in the bottom right-hand corner of the seven-segment display to denote multiple assignment. So, if an input were assigned to VCA's I and 3, the display would show "I.". The details of "hidden" VCA assignments may be seen by SEL(ect)ing the appropriate channel, and using the VCA section of the ACS to display the current VCA status.

It is also possible to set any or all of the 8 VCA faders to generate outgoing MIDI continuous controller information, to be sent via any MIDI OUT port on the system (each unit on the network has its own discrete set of MIDI ports).

An eight-character dot matrix display is housed above each of the 8 faders for name information, or to display the current MIDI continuous controller assigned to that fader. The SOLO and MUTE switches above the VCA faders will solo or mute all channels under the control of that fader.

To the right of the 8 VCA faders lies the Cross-Fader (X-FADE). This allows the user to create a smooth transition from the parameters of one cue to those of another. This transition may be automated, or performed manually by the user. Touching the X-FADEr during an automated crossfade will result in the user taking manual control of that fade.

VCA Extender Surface

Hardware

The VCA Extender surface is based on the Input Surface frame, with faders in the lower tray and a removable meterbridge in the upper section (where the ACS lives on the Input Surface).

It is intended to offer the "power user" a number of functions designed to streamline the programming and mixing of a large system. The hardware comprises:

A fader tray, containing:

- 12 VCA Faders, with associated SOLO and MUTE switches, 8-character displays for naming, and RANGE and VCA LEDs (all as per Master Surface VCA section)
- I Crossfader with associated X-FADE switch
- [NEXT], [LAST] (covered) and (Pre)VIEW buttons
- 8 Global Bank Select buttons

A centre tray, containing:

- An embedded PC
- 3.5" Disk Drive

A meter bridge, containing:

- 20 meters, 16 LEDs + peak each with an associated 4-character matrix display, a POST (amber) LED, a STEREO (GREEN) LED and a MUTE switch (with RED LED)
- Three 8-character matrix displays sited adjacent to each other, and a pair of associated function switches (with red and green LEDs)

Functional Description

Fader Tray

Broadway allows 20 VCA groups per cue. The VCA faders will function as VCAs 9-20 in the system (VCAs I-8 are present on the Master Surface faders). The crossfader is to the right of the I2 VCA faders. Operation of these faders is exactly as per the Master Surface VCAs.

The [NEXT], [LAST] (covered) and VIEW switches are to the right of the faders, and are aligned vertically. [NEXT] is at the bottom, then [LAST] (covered) and VIEW at the top. These switches are illuminated, and function as per those on the Master Surface.

Global BANK select switches operate as per those on the Master Surface.

Centre Section

The PC, mounted flush with the surface, is for use with the Broadway PC software, and sockets in the rear panel of the surface will allow for the connection of mouse/trackball and keyboard. The PC is offset to the left of the centre section to allow for the use of a mouse or small keyboard to the right. The mounting of the PC is very low-profile to allow scripts, folders etc. to be laid on top when the PC is not required.

Meterbridge

This offers 20 assignable meters, each with an associated name, pre/post-fade status and stereo LEDs. The assignments of the 20 meters is stored when a console snapshot is taken, and recalled as appropriate with each cue.

The meters may monitor signal level for any console Input, or any Graux or Matrix output. They follow the same law (and, of course, the same value!) as the meter on the channel being monitored at all times.

The pre/post led is labelled POST, and will only illuminate when the meter on the channel being monitored has been assigned to post-fade (the exception rather than the norm, keeping the number of illuminated LEDs to a minimum at any time).

The STEREO led illuminates when the channel being monitored has been assigned to be stereo. In this case, the meter shows the peak signal of both meters.

The MUTE switches below the meters will MUTE the appropriate channel. it will perform the function exactly as would the MUTE switch on the channel itself, and is directly software-linked to that switch (i.e. will be illuminated when the local channel MUTE is activated).

The 24-character display bar (i.e. 3×8 -character displays), will display the current scene name in run-time, and is used for assignment of the meters when the appropriate edit mode is selected (via the touchscreen and PC software).

In normal operation, the two function switches beside the display will operate as [NEXT]/[LAST] switches. [LAST] will be the left function switch, [NEXT] will be the right function switch. "<" and ">" respectively will be on the leftmost and rightmost extremes of the display to denote the function of the switches. In this mode, the GREEN LEDs are illuminated.

Edit Mode

The assignments of the meters may be set up in one of three ways -

- I. Via the touchscreen. A dedicated touchscreen page (on the master surface) lists the meters (1-20). The assignment of channels to meters may be performed in the same way as assignment of inputs to faders i.e., each meter number will have a name beside it the currently monitored channel. Touching the name will bring up a list of all the possible entries, and the jog wheel / screen arrows will scroll through the options until [EXECUTE] or CANCEL is pressed.
- Via the PC. Similar to the touchscreen method, but running on the PC software.
- 3. Via the 24-character display on the meterbridge. When the surface is put into EDIT mode (via the touchscreen or PC software), the leftmost display shows "XXXX" to denote that this is the meter to be edited. The 24-character display will now show the meter number (in this case "I") and the name of the currently-monitored channel. The two function keys beside the display select for editing the meter to the left or right of the current meter.

The leftmost and rightmost extremes of the display should show "<" and ">" respectively to denote the function of those switches. The RED LEDs should be illuminated when the user enters EDIT mode. When a meter is active for editing, the action of SELecting a channel (input or output) on the system will place that channel on the currently-selected meter. The assignments of channels to meters will be stored and recalled along with the rest of the cue information.

BROADWAY

Touchscreen Functions

General Syntax

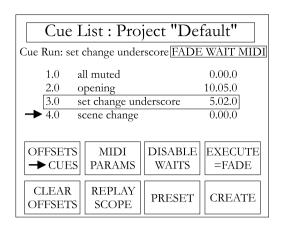
The touchscreen on Broadway contains all the information required to sceneset and configure even the largest Broadway system.

Beside the touchscreen are a number of main page switches. These present the "top-level" screens of the automation system. Within each of these pages, square, named "buttons" are drawn at the bottom of the screen. Pressing these areas on the touchscreen will "press" those buttons, the function of which varies from screen to screen. These are referred to in this document as "soft keys", and a named soft key will be referred to as [NAME], where "NAME" is the wording within the soft key.

In some pages, areas of the screen may be pressed for direct access to parameters. Once selected, some parameters are set up from a list of options presented on selection of the parameter, others by moving the jog wheel followed by touching the screen, or simply pressing the jog wheel.

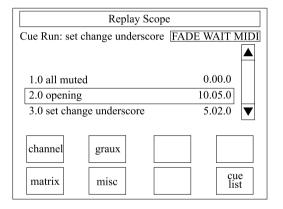
Naming functions are performed via a QWERTY keyboard, which is presented on the screen whenever appropriate.

Some of the functions described in the following pages are explained in much more detail in the section "Playing Back Cues and Mixing".



(**BOLD TEXT** denotes a hard switch beside the screen, [BRACKETS] denote touch-screen switches)

Once a performance is underway, this is the main operational page of Broadway. Pressing [REPLAY SCOPE] will bring up a page showing which elements of the console are to be recalled with automation. Once cues have been created, the user may select a "scope" for each cue independently by toggling each parameter on or off (achieved by touching the appropriate parameter). Fader positions, for example, may be eliminated from some or all cues.



The replay scope areas available for selection or deselection are:

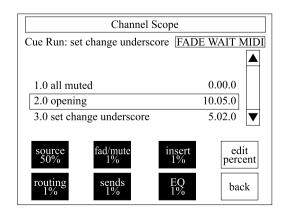
Channel: Source A/B, Phantom Power, Gain

Faders + Mutes Fader levels and mutes

Insert + Pre/Post Insert switch and Pre/post EQ switch

Routing Main Outputs & Pan
Sends Sends to all GrAuxes

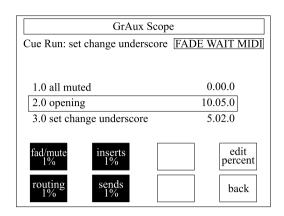
EQ settings and switches, including filters



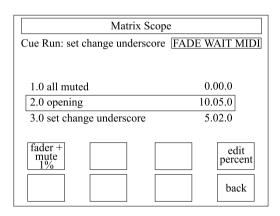
Graux: Faders + Mutes Fader levels and mutes

Inserts GrAux inserts

Routing Main Outputs & Pan
Sends Sends to all Matrices



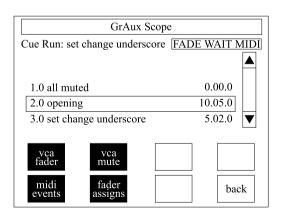
Matrix: Faders + Mutes Fader levels & mutes

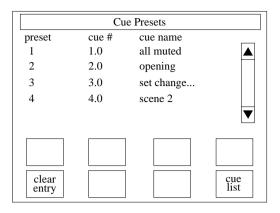


Miscellaneous: VCA Faders Fader levels

VCA Mutes Mutes

MIDI + Events MIDI messaging and eveny relays I/P Surface Assignments Assignments of inputs to faders

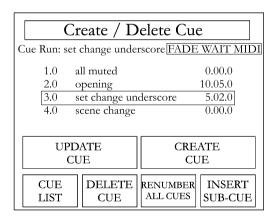




[PRESET] will bring up a list of the 8 CUE PRESET switches, which appear physically on the left hand side of the Master Surface. Against each of these entries in the list is the number and name of a cue. Pressing the cue name will bring up a list of all the cues on the system, and one of these may then be chosen for assignment to the appropriate Preset switch.

The Preset switches just recall the cue - no more, no less. They are, of course, autocancelling, and cannot be layered (just as only one cue may be recalled to the system at any time).

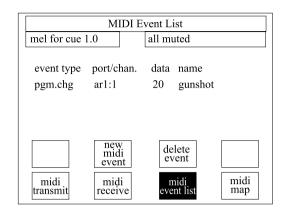
Create



This page allows the user to remove the highlighted cue permanently from the Cue List. Since this function cannot be undone, the console will ask the user to confirm deletes.

MIDI Event Lists (MEL's)

Pressing [MIDI PARAMS] on the cue list page will bring up the MIDI programming page.



The MIDI Event Lists are the heart of the MIDI control system. Each cue carries an associated MEL; if the MEL contains anything, an "M" appears to the right of the timecode in the listing on the "Cue List" page. A typical MEL would be:

CUE x.x

EVEN	T TYPE:	MIDI PORT /CHAN:	DATA.:	NAME:
I	CC	ISI:I	07	Main Volume
2	PGM	IS1:15	42	Big Rev
3	SYSEX	IS2:n/a		NEWREV.syx
4	PGM	MS:14	12	Chorus
5	NOTE	ARI:3	G#3	TRN SMPL

All surfaces and racks on the Broadway network have their own independent MIDI IN, OUT and THRU. MIDI information can be sent to any of these ports exclusively, so any Broadway system will actually support up to " $X \times I6$ " MIDI channels, where "X" is the number of network devices.

A new MIDI event may be added to a cue at any time. A requestor box will appear, showing all the possible MIDI event types available. Each event may have an associated name, which is solely for user notes, and has no MIDI function.

Pressing the NEW MIDI EVENT switch on the touchscreen brings up a callout box containing a list of all possible MIDI event types. Pressing one of these options will select that type of event, and the desk will prompt the user for an output from which to send the event. This may be any unit on the system: all surfaces and racks have their own MIDI IN, OUT and THRU.

Once the output has been selected, the system will show the new event in the list. The console still needs more information: pressing the value in the data column allows the user to set up the value of the event, i.e. CC number if the event is a continuous controller, NOTE value (including velocity) if the event is a NOTE ON, etc.

Finally, a NAME may be added to the event by pressing the entry in the NAME column.

Program Changes

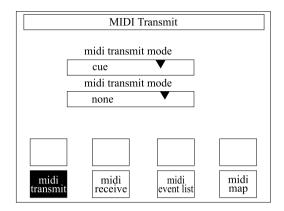
A Program change value of 0-127 may be sent from any of the units on the network. This can be useful for recalling "patch", or "memory" locations in external equipment such as reverb devices, samplers or MIDI playback units.

Notes

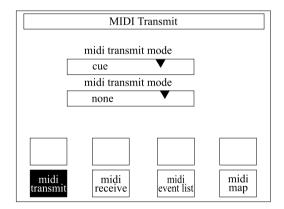
NOTE ON data types send a MIDI note value to the OUT port on the specified port and channel. The default port is that on the Master Surface (MS). The note number and velocity (shown as "real" piano keyboard note values) are specified in the NUMB. column.

Continuous Controllers

CC events may be generated by any or all of the VCA master faders. A dedicated VCA fader is required for each Continuous Controller. This can be assigned by selecting the event type with the appropriate VCA as a master, e.g. **[CC on VCA 2]**. The VCA name display blanks in this mode.



[TX] (Transmit)



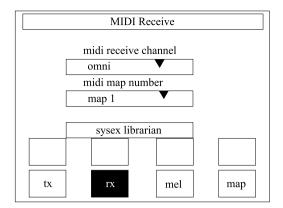
This page sets up the Transmit settings of the console. The MODE may be either **GLOBAL** or **CUE**. In **Global** mode, the console will Transmit a single Program Change (the number of which will correspond to the sequence number of the recalled cue [0-127]) from all MIDI ports each time a cue is recalled. The Program change will be sent on the channel defined in the "TRANSMIT CHANNEL" parameter ([1-16]).

Global Mode requires little setup time on the part of the user, and is useful for quickly driving a rack of MIDI gear, as it will recall a different patch on all outboard MIDI gear per cue.

If the external MIDI equipment has a Program Map, this may be used to map the incoming Program Changes to Patch locations, so the appropriate patch for each cue may be set up.

In **Cue** mode, the console will apply the MIDI Event List (MEL) associated with each cue.

[RX] (Receive)

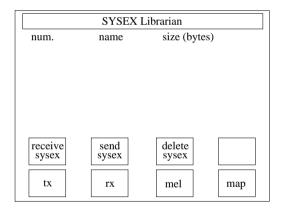


Via the RX page, it is possible to assign a receive channel for the console - i.e. the channel on which Broadway will listen for program changes from some external master device. The value may be I-16 or OMNI (any channel).

Incoming program changes may be mapped to cues via one of the program maps. These program maps are constructed in the MAP page, and Broadway can store 16 of these maps at any one time. When a new MAP number is engaged, that MAP will be automatically "USED".

System Exclusive (SYSEX)

A sub-page of [RX], the [SYSEX LIBRARIAN] allows MIDI bulk dumps to be sent from any external device into the MIDI IN. The system places this information in memory, and asks for a filename. The file extension will be ".syx".



In the MEL, the user can assign any .syx file to be sent out (i.e. associated with a particular cue). Since Sysex carries with it a unit ID number and manufacturer ident, there is no need to implement a channel assignment for this function.

Any .syx file can be selected for instantaneous transmission, should the user wish to load a whole new setup into the outboard systems.

A single .syx file may also be "tagged" to a project file, which would contain a sequential dump of each rack unit in the chain, allowing a standardised rack to be reloaded identically for each leg of a tour or for a new production of an established show.

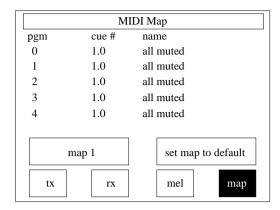
To store a SYSEX dump, connect the MIDI OUT of the device to be dumped to the MIDI IN of the Master Surface. Press [SYSEX LIBRARIAN] in the MIDI [RX] page. The screen will now show a list of the SYSEX files on the hard disk, and three additional switches. The new switches are:

[RECEIVE SYSEX] - Prepares the console for an incoming SYSEX dump. The screen shows an empty rectangle which fills to show progress, and offers an [ABORT].

[SEND SYSEX] - Sends the currently-selected file from the Hard Disk to the MIDI OUT.

[DELETE SYSEX] - Removes the selected SYSEX file from the Hard Disk. The system asks for confirmation of this procedure, as the delete function cannot be undone.

Map

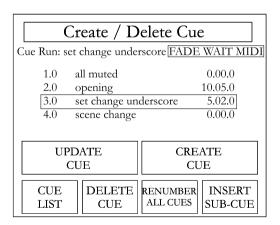


This page allows the user to define and edit a MIDI PROGRAM MAP for incoming Program Changes. This can be useful if an external device has limited MIDI implementation, and can only send fixed Program Changes per cue. These may be mapped in the Broadway to recall the desired Cue at each point.

Pressing MAP on the MEL page brings up the current remapping assignments (listed by PGM CHG number), which dictates the cue to be recalled by different incoming MIDI program changes. Pressing a cue name allows a different cue to be assigned to the appropriate PRG CHG. SET MAP TO DEFAULT will assign all cues in the cue list sequentially to PRG CHG numbers, starting at 0, and ending at "x-1", where "x" is the number of cues.

Create

This is where cues can be created, deleted and edited.



Pressing [Create It] at any time will snapshot the console position and store it as a new cue at the end of the list with a default name of "Unnamed Cue". Pressing the "name" box to the top right of the screen will bring up a keyboard, on which the cue may now be named.

Pressing [Cue List] will return the user to the CUE LIST screen.

Pressing [Update Cue] will overwrite the currently-highlighted cue with the current console setup.

See "Creating Cues" for more information on the use of this screen.

Fader Assign

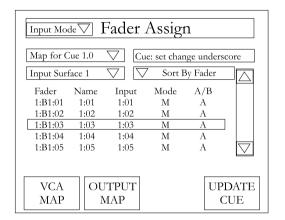
This page allows assignment of Names to Inputs, Outputs and VCA Groups, and then assignment of those items to Faders. It also allows assignment of Input Faders to VCA's.

When Fader Assign is first accessed, it will default to the "current" or "last recalled" CUE. Making changes at this point will edit the current desk setup, which may then be written to the hard disk with "UPDATE CUE" (this switch will only become available once changes have been made to the Fader Assignments).

It is possible to change the setting of other cues from within this page, but the cue to be edited **MUST** be **EXECUTE**d before editing is possible. To recall a cue from within the Fader Assign page, press the cue name at the top right of the screen, and scroll through the cue list. When the desired cue is shown, press EXECUTE. The cue will be recalled to the desk, and may then be edited. To return to the last recalled cue (i.e. the cue from which the Fader Assign page was entered) without recalling any other cue, press **CANCEL**.

Note:

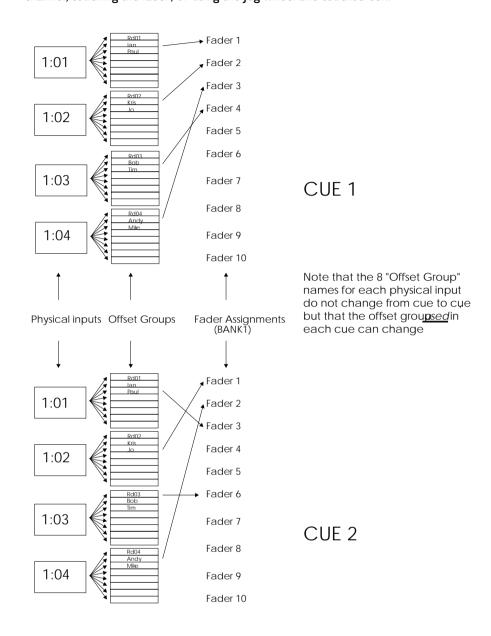
Any changes to Fader or VCA Assignments will NOT be saved until the user performs a [SAVE PROJECT] or a [SAVE AS].



There is a fundamental concept which is the key to understanding Broadway fader assignments - the fact that PHYSICAL INPUTS (i.e. rack elements) are named, and NOT the faders. Inputs are NAMED, then NAMES are assigned to FADERS. So, when assigning faders, the console will ask the user to select a fader, then give a list of all available NAMES. Every input will either have a user-defined name, or its name will default to being simply its rack and input number.

Fader / Name / Input

Pressing Fader Assign will bring up a list of all Input faders on the system, delineated by BANK. A fader may be selected for editing either by pressing SEL on that channel, touching the fader, or using the log wheel and touchscreen.



Naming Inputs

This section describes naming of inputs - if names are not required, skip directly to Assigning Faders below, as the console will simply use the default names (derived from the physical location of the input).

Each input on the Broadway system may be given up to eight names for any project, and may therefore handle up to eight different alternating sources in the same physical connector (e.g. different people using the same radio mic, etc.). This ability to handle up to eight names is relevant to a special feature of Broadway called Offset Groups, which will be explained in the Automation section below.

To name inputs, simply move the cursor to any fader on the touchscreen, and press the entry in the Input column. The system will then ask which physical input the user wishes to select, by asking for the Audio Rack number and the physical input number. It will then present a list of the eight names associated with that input. Turn and press the jog wheel, or touch the screen to select one of these.

At this point, the user may either edit the current names, or create new ones. Selecting one of the eight names will bring that name into the edit box at the bottom of the menu. Pressing the edit box will bring up a keyboard via which a new name may be entered. Names may be up to four characters in length. Once the appropriate names have been entered for that input, selecting one of the names again will select that name (and therefore the associated input) for use on the current fader.

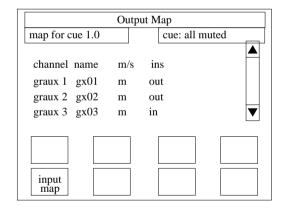
Repeating this process for all inputs on the system will give names to every physical input. Although time-consuming in the short-term, this process will make fader reassignments much easier in the long term, since the user can forget completely about physical inputs, and just refer to inputs by their name.

Bear in mind that the process of naming inputs may be carried out via a single fader. Once an input has been named, simply press the input column again on the same fader, and select a different input.

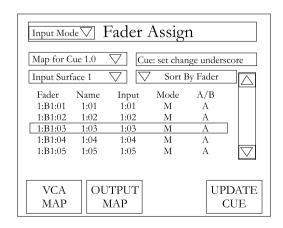
Naming Outputs

The naming of outputs is performed in a similar manner to the naming of inputs, except that the list is always in the order: GrAux I through GrAux 32. This is because the GrAuxes cannot be reassigned to different faders.

Pressing the current name of the required GrAux will bring up a list of the 8 offset groups associated with that GrAux. A different offset group may be selected at that time by rotating the jog wheel, then pressing the required offset group, which may optionally be renamed.



Assigning Faders



Once a fader has been chosen (selected on the touchscreen), pressing the FADER column will bring up a list of all names available on the system, i.e. up to 8 per input. The Jog wheel will scroll through these, until it is pressed down, at which point the currently-selected name is put onto that fader. If no user names have been defined, the list of names will simply be a list of all the inputs, sorted numerically.

If the inputs have been named in advance (see above), then the process of assigning inputs to faders is as simple as selecting a name from a list, and the physical location of the input is no longer an issue (as the console "knows" this information already).

Pressing the "NAME" entry will bring up a list of the 8 Offset Groups/Names associated with the current input. A different name may be selected from the list by touching the screen.

Stereo Channels

It is possible to link two adjacent mono channels in an input rack for use as a stereo pair.

This is achieved by pressing the "m" in the mode column of the appropriate channel, which will then become an "s" denoting stereo. The touchscreen will then show the "linked" partner with the use of brackets, so the same two channels appear on one fader. The first input shown is the "master" input for the stereo pair, and will appear on the Input surface. The second input shown (in brackets) is the slave, and, if that input is assigned to an Input surface location, it will not show on the fader tray, and is effectively a "wasted" fader assignment, as the Master of the pair will allow all the required functionality of the channel.

All ACS functions performed on the Master channel are automatically copied to its partner, except for the PAN control. When a channel is defined as stereo, the PAN on the ACS becomes a BALANCE control, and dictates the relative levels derived from each of the mono inputs. Since the SLAVE channel cannot be assigned to a fader, it cannot be SELected.

Routing to a GrAux from a stereo channel results in the stereo pair being sent in mono to the GrAux with a 3dB drop in each leg to maintain the desired level. Routing to the Main outputs will result in the Left leg of the pair being sent to OPI and 4, and in the Right leg being sent to OP2 and 5.

Assigning VCAs

The term "VCA" is used because the faders on the Master section operate in the same way as master VCA group faders on a conventional desk. The VCA control fader works in a conventional way, and has a normal fader scale marked in dBs. The position of the VCA fader is added to the slave faders. The slaves themselves will not move, but the value of the VCA Master Fader is added to all appropriate slaves. Assigning an input to a VCA is therefore like adding a second fader into the channel path in series with the first, but the maximum gain applicable from fader combinations cannot exceed + I0dB (see below).

The VCA Master Faders are motorised, and their position may be stored in the automation system along with everything else. They are scoped out of recall by default.

Just as for other parts of the system, the VCA faders may be removed from recall via the "Scope" function (see "Replay Scope" below). These faders may then be viewed as "Live" offsets of stored fader positions. The advantage of VCA groups over ordinary audio groups is that changes to VCA levels will affect post-fader sends from any slave input channels, and the MUTE and SOLO switches are linked in logic to those on the slave channels.

See "Automation" below for information about how VCA fader positions affect sceneset recall.

There are 8 VCA control faders, but the system supports up to 20 control groups per cue. The standard Master Surface alone can only handle 8 control groups. The full 20 control groups can only be controlled via the VCA mode on the Input Surface, or via the optional VCA Extender Surface.

VCA assignments are performed in the Fader Assign Page. Pressing [VCA] will bring up a list of the current VCA groups with all slave faders listed beside each. Pressing the slave list on any VCA will bring up a list of all possible slaves which are the CURRENT offset groups on each channel. The Jog wheel will navigate around the list of available slaves, and pressing the wheel at any point will toggle the current slave on or off for the selected VCA.

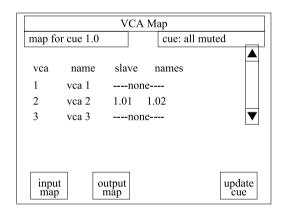
Pressing DONE will return to the VCA assignment screen. At this point, the new settings have been assigned to the cue, but not to the desk. It is necessary to go to the CUE LIST screen, move the cursor to the last recalled (or "current") cue, and press [EXECUTE] to recall the cue along with its new VCA assignments. This will bring the new assignments to the desk.

It would be a very slow process if a channel had to be assigned to a VCA in a number of existing cues. Broadway gets around this problem with automatic assignments.

If a channel is assigned to a VCA in a particular cue, then this assignment will automatically be written to all subsequent cues, <u>until it reaches a cue in which another specific assignment has been made.</u>

For example, let us presume that the "SNRE" channel has been assigned to VCA 2 in Cue 5. Suppose that the same channel needs to be under the control of VCA I in cues I to 4 inclusive. Adding "SNRE" to the slave list of VCA I in Cue I will now have that effect, as the system will automatically update the entries for Cues 2, 3 and 4 (because there were no previous assignments there). It will, however, stop at Cue 5, because it recognises that an assignment was previously made at this Cue.

On the other hand, if "SNRE" was only required to be under the control of VCA I in Cue I, then it will need to be removed from Cue 2 manually, which setting (i.e. off) will now be written through Cues 2, 3 and 4 automatically.



VCA Master faders may be given names of up to eight characters in length by pressing the VCA name area, and entering a name via the QWERTY keyboard.

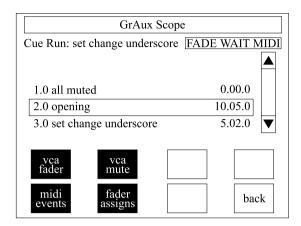
VCA assignments are unique to each Offset Group. That is, if Offset Group I of an input is assigned to a VCA in a cue, and the second Offset Group is then chosen instead, that channel will no longer be under VCA control. At this point, the slave channel must be reselected via the VCA MAP page. Once the list of possible slaves appears for a VCA, the name of the second Offset Group for that input should now be offered in place of the first option, as this is now the CURRENT Offset Group for that input.

VCA LEDs and Switches

The MUTE switch above the VCA will trigger the MUTEs on all slave channels. When the MUTE is removed from the VCA master fader, the slave channels will be returned to their status before the VCA MUTE was triggered (i.e. MUTEd if they were muted, open if they were not).

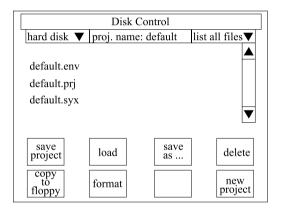
Note that channels muted via a VCA master fader **cannot** be unmuted locally on the channel. Muting a slave from a VCA master effectively locks out the local channel MUTE switch (above the SELect switch).

Setup contains all global configuration parameters, along with user preferences and Disk handling.



Disk Handling

[DISK] brings up the disk access page.



File Types

There are three types of disk file: .PRJ, .SYX and .ENV.

.PRJ files are the main project files, and contain all the automation da

.SYX files are System Exclusive files. These are manufacturer-exclusive MIDI files which allow a user to dump the memory contents (parameter settings) of a MIDI device over the MIDI link. A single System Exclusive dump may be attached to any project, and will be automatically sent to the MIDI ports when the main associated project (.PRJ file) is loaded.

.ENV files contain the user preferences. Using these files, the operator may save their preferred operating environment, including touchscreen configuration preferences, to a floppy disk, and carry it with them. Whenever using any Broadway in the future, it will then be possible to load their own working environment onto the system without affecting the stored automation data.

Saving

[SAVE AS] allows the user to save the current console setup as a new project file. The console will prompt for a file name, which may be up to eight characters in length. Note that the current file type will be taken into account when saving. Ensure that .PRJ (project file) is selected if the whole project is to be saved. [SAVE PROJECT] will save the current desk project to disk, ensuring that all touchscreen-programmed parameters are saved to the hard disk. Cue Data (i.e. channel parameters but **NOT** VCA / Surface assignments) is stored automatically whenever a cue is created or updated, but a SAVE or a [SAVE AS] (see above) must be performed if VCA / Surface assignments are to be permanently stored.

Loading

[LOAD] will present a list of all project files on the current drive. The current drive can be changed by selecting the box containing the drive letter (C: or A:) at the top of the screen. C: is the internal hard disk. A: is the floppy drive.

Formatting

Disks may be formatted using the [FORMAT] command. **TREAT THIS FUNC-TION WITH GREAT CARE**. Formatting will erase all data on the selected disk, and this information is irretrievable. Particular care should be taken when formatting Floppy disks - always ensure that the A: drive is selected, and that the C: drive is not about to be erroneously formatted, as this could be very disappointing indeed.

Creating A New Project

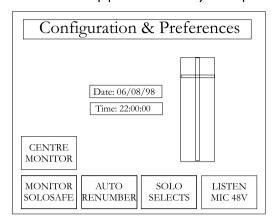
The [NEW PROJECT] command will generate a new .PRJ file on the Hard Disk, which may then be named.

Copy to Floppy

Copy to Floppy will take the hilighted project and copy it to the Floppy Drive for backup and later retrieval. When the A:\ (floppy) drive is selected as the current drive, this switch will become simply "COPY", and will copy the hilighted project from the Floppy Drive to the Hard Disk.

Note that Projects cannot be run directly from the A:\ (floppy) drive - they must first be copied to the C:\ drive (hard disk).

[CONFIG] allows the user to set up preferences for system operation.



MONITOR SOLOSAFE

Pressing any SOLO switch on Broadway will, by default, over-ride the Headphones and Monitor Output with the SOLO signal.

However, in some applications, it may be desirable for the Monitors to remain unaffected by SOLOs. When **MONITOR SOLOSAFE** is activated, the console will over-ride ONLY the Headphones with the SOLO signal, and the Monitor path will pass as before.

SOLO SELECTS

This feature is a one-way link between the SOLO switch and the SEL(ect) switch. Usually, the SOLO and SEL(ect) functions on Broadway are discrete from one another, but when **SOLO SELECTS** is activated, pressing the SOLO switch on a channel will trigger the SEL(ect) switch on that channel also, but pressing the SEL switch will NOT trigger the SOLO.

This can be useful when SOLOing the audio for troubleshooting purposes, as the local ACS is automatically switched to monitor that channel without a second keypress.

Auto Renumber

[Auto Renumber] will renumber the whole scene list whenever a cue is inserted into the list (see "Renumbering" in the Automation section below). This means that the user need not worry about running out of "insertable" cue availability, and is useful during pre-production, where precise cue numbering is usually not as critical as setting up some starting points.

CENTRE MONITOR

The CENTRE MONITOR switch allows true LCR Monitoring to be performed. It has an effect only when the Monitor Source switches on the Master Surface are set to monitor OP1, OP2 and OP3.

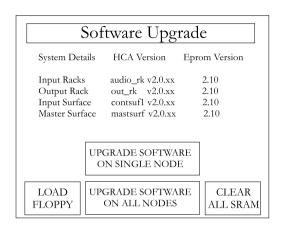
In usual operation, when OP1/2/3 are selected as the Monitor source, OP3 is sent to both Left and Right Monitor outputs.

However, when CENTRE MONITOR is selected (and OP1/2/3 is still set as the Monitor source), program material sent to the Centre buss (OP3) is directed to the CRM C(entre) output on the rear of the Master Surface.

Updating Software (See Section 9 - Software Upgrades)

The [Software Update] page allows the user to load new software releases from floppy disks onto the master surface (via the integrated 3.5" floppy drive). These may then be downloaded over the network to the other Broadway units (or "nodes").

Pressing [Software Update] brings up the Update screen. This shows the unit type, current version number and network ID of all units found on the network.



Loading new software is a two-stage process. The software has to be copied to the Master Surface, then distributed through the network. When new software is provided on floppy disk(s), placing the disk(s) in the Master Surface drive, and pressing [Load Floppy] will copy all of the data over to the Master Surface. Messages on the screen will show what the Master Surface has found on the disk. There are different files for each surface and rack type, and system data.

The user may then choose to send the appropriate new version to one unit on the network (using the cursor followed by [Upgrade Software On Single Node]), or upgrade ALL units at the same time (by pressing [Upgrade All Nodes]). After this is complete the console should have its SRAM cleared and be rebooted (see below).

Clearing SRAM

[Clear All SRAM] removes **all** data from the memory of **all** network units. It does **not** format the hard disk. Broadway units have battery-backed static ram, which is memory that survives powering down. This means that when a network unit is switched on, it already "knows" what it is, and what it expects to find on the network.Before rebooting after a new software version is loaded, the SRAM should be cleared in all devices using [Clear All SRAM], to ensure that the new software is loaded onto a "clean" system.

Rebooting After SRAM Clear

After an SRAM clear, the various parts of the console must be started up IN ORDER if the system is to boot successfully. The order is:

First Input Rack

Second Input Rack (if installed)

Third Input Rack (if installed)

Output Rack

First Input Surface

Second Input Surface (if installed)

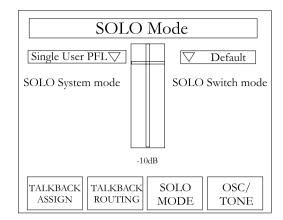
Third Input Surface (if installed)

Master Surface

VCA Extender Surface (if installed)

At least 30 seconds should be left between starting each unit and the next.

This gives access to the Solo, Talkback and Oscillator parameters of the desk.



Solo, Monitoring and Headphones

Solo signals will be sent to the Monitors and the Headphone system, dependent upon the settings of the Monitor and Headphones parameters described below in "Monitors" and "Headphones". Solo overrides Monitors and Headphones.

- The Solo system may operate in one of a number of modes:
- AFL (After-Fade Listen Solo picked up after the fader for the associated channel)
- PFL (Pre-Fade Listen Solo picked up before the fader for the associated channel)
- SOLO IN PLACE (see below)
- DIM SOLO IN PLACE (see below)

Solo Switches

In addition to the various Solo modes, there are three types of SOLO switch operation, which define how SOLO switches on the control surfaces interact. The switch operation modes are:

Momentary (Only operates whilst switch is held)

Latching (Additive)

Interlocking (Automatically cancels any previous SOLO selection)

SOLO IN PLACE (SIP) operation

In this mode, when the [SOLO] button on an input is pressed, all Input channels except for the soloed one are muted. SIP is not possible on GrAux and Matrix channels: the SOLO switches on Graux and Matrix outputs are disabled when SIP mode is selected.

Rather than muting the other channels, the SIP function may optionally DIM those channels (DIM SOLO IN PLACE mode). The attenuation applied is variable, and set in the SOLO/TB touchscreen pages, using the slider in the middle of the page.

Monitors

On the Master surface, there are four Main Output monitor switches - OP1, OP2, OP3, OP4/5. In addition to these, there are four Intercancelling main monitor sources - MONO, EXT1, EXT2 and Listen Mic. When any of the latter four is selected, any other monitor switches selected will be cleared.

The OP switches operate slightly differently. OPI and OP2 are linked, and cannot be independently selected. OP3 may be toggled independently, with the results shown in the table below.

Switch	L+R	L+R		
	L	R		
1/2	1	2		
1/2/3	I + 3	2+3		
4/5	4	5		
MONO	1,2,3,4,5	1,2,3,4,5		
EXTI	EXTIL	EXTIR		
EXT2	EXT2L	EXT2R		
LSTN MIC	LISTEN	LISTEN		
	MIC L	MIC R		

Monitor Safe

The Monitor Safe switch on this screen protects the Monitor outputs from SOLO selection (which would usually override the Monitoring). The SOLO signal will still appear on the PFL and Headphone outputs.

Headphones

The PHONES I/4" TRS Jack is a stereo amplified headphone level output. PFL I, 2 and 3 are stereo, and appear on 5-pin Neutrik XLR line level outputs.

When no SOLO's are selected on the system, the "default" signal carried by the phones will be whatever is selected via the front panel switches.

The switch laws are as per the monitor in L+R mode (see above), with the exception that the Headphones can also be selected to Follow Monitor. If this is selected, then the headphone source selection will follow the selection made for the monitors.

The I/4" TRS "PHONES" socket and PFL I will carry the default signal (i.e. whatever is selected on the PHONES switch panel on the Master surface), until such time as a SOLO is pressed, or return talkback comes into the console.

When a SOLO is pressed, the mono / stereo PFL/AFL signal is fed to the PHONES and to PFLI.

Talkback

There are a number of talkback possibilities on Broadway. The switches on the upper part of the Master Surface control the talkback from the console - talkback into the console, or "return talkback" is automatically routed into the primary headphones and the monitors.

Input to the talkback system is via the female XLR connections on the front and rear of the Master Surface. These connections are linked, and it is therefore only a matter of convenience which connector is used.

All talkback switches are momentary - that is, they will only be armed while they are held.

EXT TALKBACK routes the talkback signal to the EXT TB OUT on the output rack. When this switch is depressed, any signal appearing at the EXT TB IN on the output rack will be sent to the monitors and headphones.

MON DESK routes the talkback signal to the MON TB OUT on the output rack. When this switch is depressed, any signal appearing at the MON TB IN on the output rack will be sent to the monitors and headphones.

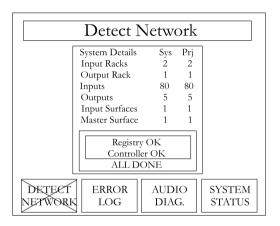
TALK TO RACK routes the talkback signal to the rack headphones, the jack for which appears on the front panel of the output rack.

TALK TO BUS routes the talkback signal to any selected Graux, Main or Matrix outputs. Outputs need to be armed for this function to operate. This is done by pressing the T/B ASSN function switch in the centre of the Master Surface, then pressing the FUNC switch on the required outputs so that the LED is illuminated. No return Talkback is activated with this function - it is output only.

Diagnostics

Detect Network

rently on the network.



The **DETECT NETWORK** page triggers a complete network scan, and reports its findings to the user. This page is useful for two reasons: it allows the user to check that all devices are correctly detected by the network, and also allows the user to compare current System size with the size of the loaded Project.

The screen is divided into three columns - **System Details**, **System**, and **Project**.

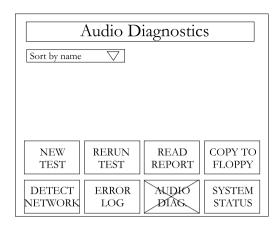
System Details lists the item categories which the console is expecting to find. The **System** column shows the quantity of each component FOUND running cur-

Project shows the quantity of each component present WHEN THE PROJECT WAS CREATED - that is, what the console is EXPECTING to find to run this Project as it was intended to be run.

Although the console will, of course, run a Project with fewer CONTROL surfaces on the system than were present upon Project creation, it will NOT play back a Project if the AUDIO configuration is not IDENTICAL to the system on which it was created.

So, for example, a 40 Input, 20 Matrix, 2 Input Surface Project will run fine on a 40 Input, 20 Matrix, 1 Input Surface System (because only the Control Surface elements have changed), but NOT on a 60 input, 20 Matrix, 1 Input Surface System.

Audio Diagnostics



The Audio Diagnostics page allows the user to configure the console to route audio through any or all Inputs to any or all Outputs (including Matrix Outputs where appropriate), and to report its findings as a file which can be copied to Floppy for Email to Soundcraft or simply for reference.

In addition to the audio continuity, Pre/Post Fader send levels, EQ performance and insert points may all be included in the test.

NB: Note that in order to perform these tests, Broadway routes the oscillator on each input card through the channel audio, onto the busses, and out of the GrAux and Matrix outputs. It is highly recommended that all amplification downstream of Broadway (house amps, monitoring amps, etc.) be muted for the duration of the audio test.

The system will store not only the results of any Audio Tests run, but also the Configuration of each test, so that the same check may be easily rerun as many times as required. This is particularly handy when troubleshooting, as it allows the operator to implement a fix for the problem at hand, then rerun the original test and compare the results to measure the effectiveness or otherwise of the solution.

NEW TEST begins the process of creating a new audio check, and brings up the necessary dialogue boxes which request information from the user about which inputs and outputs are to be tested.

RERUN TEST will restart an existing test, and will store the results with the same name as before, but with a new timestamp.

READ REPORT brings up the report file for the selected Test.

COPY TO FLOPPY will copy the report file for the selected test from the Master Surface to the Floppy Drive.

Creating a New Test

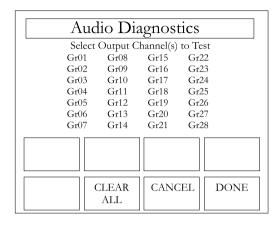
Pressing NEW TEST brings up the Input dialogue. This allows selection of the Input Channels to be tested.

Audio Diagnostics					
S	Select Input Channel(s) to Test				
	1:01 1:08 1:15 1:22 1:02 1:09 1:16 1:23 1:03 1:10 1:17 1:24 1:04 1:11 1:18 1:25 1:05 1:12 1:19 1:26 1:06 1:13 1:20 1:27 1:07 1:14 1:21 1:28				
		EAR LL	CANC	EEL [DONE

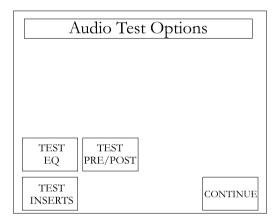
CLEAR ALL will remove all highlights, Cancel returns to the AUDIO DIAGNOSTICS page.

When the required Inputs have been selected, pressing DONE will step on to the Output Dialogue.

Select the required Outputs using the same method as the Input dialogue.



Pressing DONE when the required Outputs have been selected will bring up the AUDIO TEST OPTIONS page.



At this point, the user may decide to include in the test or ignore the EQ, Inserts and Pre/Post-fade sends. Pressing any of these options will toggle them into / out of the test. Pressing CONFIRM will bring up the QWERTY keyboard for naming of the Test.

The user may now name the Test with any combination of characters (e.g. "Test IP3&4 to all OPs"), and then OK will run the test, once the User has answered "YES" to the "Are you sure" dialogue.

Dual Network

Although Broadway requires just a single Ethernet network to operate, the console may actually be connected via 2 discrete parallel networks - A and B. Only one will actually be operating at any given time, with the other standing by as a backup. The console continuously polls the backup network even though main operation is via the primary network.

NB: at this time, the PSU sensing is not active. The screen will ALWAYS therefore show PSU I & 2 as "ok", but that is not necessarily the status of the PSU.

If the console detects a problem of any kind on the current network, it will automatically switch to using the backup network. The user will be alerted to this by way of a flashing display on the CUE LIST and CREATE pages. The words "Check" and "Status" will appear in the top left and top right areas of the touchscreen respectively. Note that at this point the user may choose to ignore the warning and continue to operate - the console will run as normal, but the warning will remain visible until the user selects DIAG then SYSTEM ERRORS.

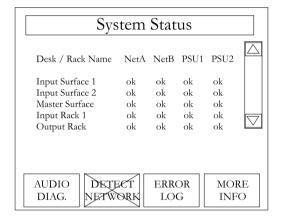


Figure I shows the SYSTEM STATUS screen. In this example, all nodes are functioning correctly. The console is aware of various types of network failure:

- An individual port failing on a single node (i.e. one port down on one node)
- An individual node failing (i.e. both Net A and Net B down on one node)
- A whole network path (i.e. A or B ports on all nodes) going down; usually because the 50(termination has failed
- Multiple nodes failing (both ports) on either network
- Multiple nodes failing (single port) on either network

EXAMPLES:

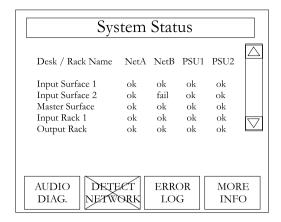


Figure 2 - One port on one node is down; probably a hardware problem with that port

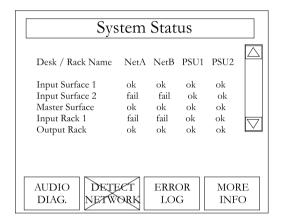


Figure 3 - Both ports on one node are down; the CPU in that node has failed to respond for some reason

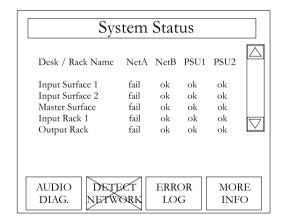


Figure 4 - One leg of the Dual network has failed completely; probably loss of termination at one end, or a short circuit in the cabling

ACTIVITY INDICATORS

LED activity: each port on every node of Broadway has two LEDs - one red (Status), and one green (Data).

A flickering green LED denotes that Data is being passed through the port.

The red LED is a Status indicator. Flashing illumination denotes that the associated port is in standby as a backup, and is functioning correctly. Remember that the console continuously polls the backup network even though main opration is via the primary network.

Regular polling means that the the desk can alert the user to a failure in the backup network before it is required (and fails...) in an emergency. Steady illumination of the red LED denotes failure of that network channel. If the console detects a failure on a channel (A or B), then it will steadily illuminate the red LED for that port on ALL nodes. So, even if it is a hardware problem at the rack end of the network which has caused the channel to go short circuit and fail, the surfaces will show failure on that port as well - if ONE fails, they ALL show failure.

To summarise:

Green LED A	Red LED A	Green LED B	Red LED B	Status
Flashing	Off	Off	Flashing	Node running correctly. A is primary network channel, backup channel is responding to polling requests.
Off	Flashing	Flashing	Off	Node running correctly. B is primary network channel, backup channel is responding to polling requests.
Flashing	Off	Off	On	Node running successfully on primary network A, but the backup channel has failed, and will not take over operation correctly in the event of failure of the primary network
Off	On	Off	On	Node is not communicating successfully on either network channel, and will therefore not respond to or transmit any control data
Any Steady State	Any Steady State	Any Steady State	Any Steady State	Node is not communicating successfully on either network channel, and will therefore not respond to or transmit any control data

BROADWAY

6 Automation

6. l **BROADWAY Automation**

Overview

All audio parameters on Broadway, including switches, rotaries, faders and text displays, are included in the snapshot automation system.

"View" parameters, such as the current BANK, SEL settings, solo's etc. are not stored or recalled. VCA Master Fader positions are stored on a per-cue basis, as are the assignments of faders to VCA's.

VCA master faders are included in the automation system, but may be "scoped" out, effectively leaving them in manual mode. The idea is that the relative balance between a group of faders (perhaps a group of radio microphones) can be set up on the input faders, and these may all be "moved" together, retaining the relative balance, via a single fader - the VCA master.

The dB offset applied via the VCA master will almost certainly be different from day to day. When a scene is recalled, therefore, the input faders are returned to their stored values (as long as their "scope" is switched in), and the appropriate VCA master fader offsets are applied to the gain for each of the inputs slaved to that VCA.

Terminology

Complete console snapshots are stored as "Cues". Cues may be named, and may have MIDI and Event data associated with them. Cues may be arranged into a list called, unsurprisingly, the "Cue List". A complete cue list may be stored as a named Project on the internal hard disk, or onto a Floppy Disk.

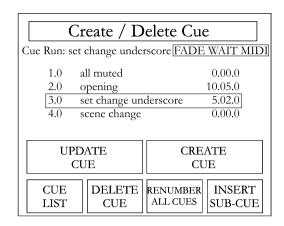
The highlighted row in the cue list is marked with the cursor - an inverted bar. The cursor may be moved around the screen via the jog wheel, or, in some cases, by touching areas of the screen. The jog wheel includes a push-switch for selections and editing.

Below the touchscreen are a number of "transport" switches which are used primarily for moving between cues. Only those switches which are illuminated are available for selection.

BROADWAY Automation 6.3

Creating Cues

Once the console has been set up as desired, pressing **Make Cue** will bring up the cue creation window.



At the top left of the screen is a selector which allows the user to select "ADD MAIN CUE" or "INSERT SUB CUE".

To **add a new cue** to the end of the list, check that ADD MAIN CUE is selected, and press [Create It]

To **insert a cue** at the current cursor position, select INSERT SUB CUE, and press [Create It]. A new cue will be inserted after the cursor position, and allocated an appropriate sequence number half way between the two numbers either side. So, inserting a cue between cues 3.0 and 4.0 will generate a cue in between those two, numbered 3.5. Insertion of additional cues will follow the same numbering principle, as the system attempts to find a suitable half-way point to the nearest single decimal place. Once no further legal sequence numbers remain in a given space, it is advisable to renumber the list before adding more cues. See Renumbering below.

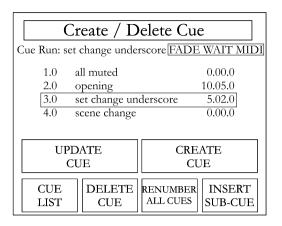
To **overwrite the highlighted cue** with the current console status, press [Update Cue]

New cues are created with a default name of "Unnamed Cue", which may be edited by touching the name box in the top right area of the touchscreen.

At the bottom of the screen are [Renumber All Cues] and [Cue List].

[Cue List] returns the user to the Cue List screen, as would pressing the **Cue List** hard switch.

Renumbering



[Renumber All Cues] assigns integer sequence numbers to every cue from 1.0 to X.0, where X is the number of cues.

Once part of a cue list has reached saturation (i.e. too many inserted cues), it is advisable to renumber the list before more cues can be added. Renumber will do just this, and will ask for confirmation to avoid unfortunate mishaps.

It might also be useful to renumber a list once rehearsal or setup time is over, and the project has settled down.

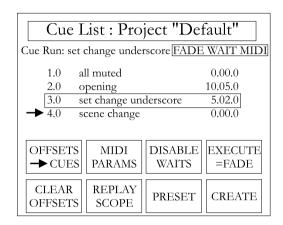
The user may choose to renumber cues automatically when insertions or deletions are made. See "Auto Renumber" (5.4.3.2) above.

BROADWAY Automation 6.5

Replaying Cues

In normal operation, the Cursor Bar will lie on the "current", or "last recalled", cue. The **[NEXT]** and **[LAST]** switches will recall the next and last cues respectively, and the cursor will follow the current cue.

The jog wheel will scroll through the available cues for the current project. When any cue is highlighted, the **[EXECUTE]** switch will flash. This indicates that **[EXECUTE]** will recall the highlighted cue to the desk, and that cue will become the current cue. In this way, it is possible to recall cues out of sequence, especially useful in rehearsal or unpredictable live set environments.



Crossfades

Although cues are normally recalled to the console immediately, Broadway is also able to crossfade settings from the current desk state into the next cue. This can be useful when smooth transitions are required (for example a fadeout of a sound effect and a fade in of some live Microphones).

Crossfade will take the current position of all faders, and move them to the settings stored in the upcoming cue over the defined length of the crossfade. Since the time is defined globally for the cue, all faders will take the same time to fade. Therefore channels for which the adjustment in the new cue is greater will "move" faster than those with small adjustments.

Note that only Fader levels are crossfaded - all EQ settings, switches, filters etc. are snapshot as usual, but the POINT in the crossfade at which the various elements are switched may be reconfigured. Note also that although the audio is crossfading, the physical faders on the surfaces DO NOT move during the crossfade - they will move into position only once the crossfade is complete.

Crossfades can be achieved in one of two ways: timed crossfade, or manual crossfade.

Timed Crossfade

The user may define a crossfade time for each cue. The time is edited by pressing the area in the "Fade" column on the Cue List screen for the associated cue. Note that this time relates to the duration over which movement INTO that cue will be performed. It has no effect on the crossfade time into the next cue.

By default, Broadway will ignore the crossfade time when executing cues out of sequence with the jog wheel and EXECUTE switch (for example when programming during a break). However, the EXECUTE=FADE toggle switch on the Cue List page will result in crossfades being performed even when cues are recalled via the EXECUTE switch.

When a crossfade is underway, the XFADE switch to the left of the Touchscreen panel flashes, and the XFADE fader begins to move from 0% (the bottom of its travel) to 100% (top) to show the percentage of the crossfade completed.

At any point during the fade process, the user may interrupt the process by touching the XFADE fader. This will halt the process, and the user may then complete the fade manually by moving the fader towards the top of its travel. Of course, the fader may equally be moved downwards to reverse part of the fade, but at some point it must either be parked at the top, or the crossfade cancelled.

The user may also touch any channel fader to drop that channel out of the fade, and return it to manual operation.

When the XFADE fader reaches the top of its travel, the cue is deemed to be recalled, and the touchscreen display will be updated to show this.

To remove a FADE time from a cue, enter the edit box as usual, but type "0.00".

Manual Crossfades

A manual crossfade is begun simply by pressing the XFADE switch. The switch will illuminate to show that a crossfade is active.

The XFADE fader will snap to the 0% mark, and the user may then use the fader to perform the crossfade at the desired speed.

Switch points

The point at which switched functions (i.e. anything other than Faders) may be configured per function block, per cue. This is achieved using an extension of the REPLAY SCOPE functions.

Pressing REPLAY SCOPE will, as usual, bring up a list of the available scope blocks. Choosing one of these blocks will show the list of available scope choices, each of which has an associated percentage.

The percentage will show either:

1% Switched as soon as the crossfade starts

50% Switched half way through the crossfade

100% Switched at the end of the crossfade

To edit the percentage of a particular scope, simply turn on the EDIT PERCENT toggle switch, and press the appropriate scope to cycle that scope through the percentage options.

WAIT

The console will usually only recall cues when the User manually activates a recall, or the system receives a valid incoming MIDI Program Change.

However, cues may also follow on automatically from each other. This is achieved using a function called "WAIT".

The WAIT time is listed alongside the Fade time for a cue, and is edited in the same way (just press the space in the WAIT column for the appropriate cue). WAIT time is defined as the time that the associated cue will wait before automatically recalling the next cue

So, if two cues exist - A and B, putting a wait time of 3 seconds against cue A, then recalling Cue A will result in Cue A being recalled immediately, and 3 seconds later Cue B will be automatically recalled.

Note that a wait time of "0.00" will result in the next cue being recalled IMMEDIATE-LY. So, to produce a fast sequence of automatic Cues (for example for a series of chained sound effects cues with MIDI Note Transmission), just give all the cues WAIT values of "0.00".

To remove a wait time from a cue, enter the edit box as usual, but type "-.--". Remember that "0.00" is not "WAIT OFF", but "IMMEDIATE RECALL".

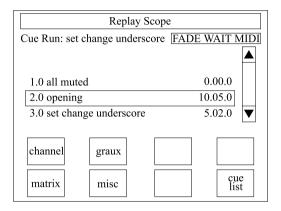
BROADWAY Automation 6.7

Replay Scope

Whenever a cue is created or updated, the console stores every audio parameter to the automation system. However, upon recall, the user may define which areas of the console should be recalled with each scene.

There is a useful "copying" function which can be achieved with REPLAY SCOPE (see section 7.4 "Selectively Copying Data Between Cues".

Pressing [Replay Scope] will replace the touchscreen switches at the bottom of the Cue List screen with a list of areas of the console, most of which are highlighted in the default state. Each of these areas (EQ, FADERS, MUTES, etc.) is presented as a "toggle" switch - that is, pressing the switch once will deselect that area of the console, pressing it again will reselect it.



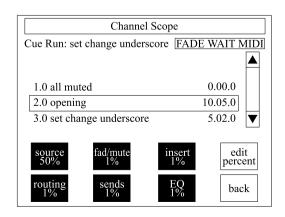
When any cue is recalled, the Scope settings associated with that cue will define the areas of the console to be recalled in that scene.

By scrolling through the cue list with the jog wheel while [Replay Scope] is selected, the Replay Scope of each cue may be set individually. It might be useful, for example, to "ignore" fader positions for ten scenes in a row to allow for live adjustments, then include the faders in the eleventh cue, to reset them to a required absolute position.

The replay scope areas available for selection or deselection are:

Channel:

Source
A/B, Phantom Power, Gain
Faders + Mutes
Insert + Pre/Post
Routing
Sends
Fader levels and mutes
Insert switch and Pre/post EQ switch
Main Outputs & Pan
Sends to all GrAuxes
EQ Settings and switches, including



filters

Graux: Faders + Mutes Fader levels and mutes

Inserts GrAux inserts

Routing Main Outputs & Pan
Sends Sends to all Matrices

GrAux Scope				
Cue Run: set	change undersco	re FADE WAIT MIDI		
1.0 all mute	ed	0.00.0		
2.0 opening	;	10.05.0		
3.0 set chan	ge underscore	5.02.0		
fad/mute 1% routing 1%	inserts 1% sends 1%	edit percent back		

Matrix: Faders + Mutes Fader levels & mutes

Matrix Scope				
Cue Run: set change underscore FADE WAIT MIDI				
1.0 all muted	0.00.0			
2.0 opening	10.05.0			
3.0 set change underscore	5.02.0			
fader + mute 1%	edit percent back			

Miscellaneous: VCA Faders Fader levels

VCA Mutes Mutes

MIDI + Events MIDI messaging and event relays

I/P Surface Assignments Assignments of inputs

GrAux Scope				
Cue Run: se	t change under	score FADE WA	AIT MIDI	
1.0 all mu	ted	0.0	0.00	
2.0 openir	ıg	10.0	05.0	
3.0 set cha	inge underscore	e 5.0	5.02.0	
vca fader	vca mute			
midi events	i/p surf assigns		back	

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Preview

To the right of **[LAST]** is **[VIEW]**. This is a latching mode switch. When selected, pre**[VIEW]** mode will allow the user to recall cues to the surfaces whilst leaving the rack audio as it was when **[VIEW]** was first pressed.

This can be useful if the operator needs to recall upcoming cues to check their contents, whilst a fairly static scene is being performed on stage. There will be no changes to the audio, but the surfaces will reset just as they would normally.

It is also possible to edit and store data when in pre**[VIEW]** mode. The syntax is just as when in normal operation, except the audio will not follow the surface control movements.

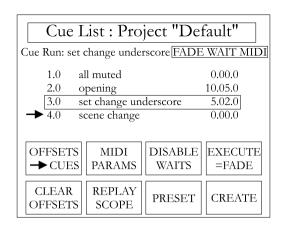
Coming out of [VIEW] mode will return the surfaces to the settings of the audio

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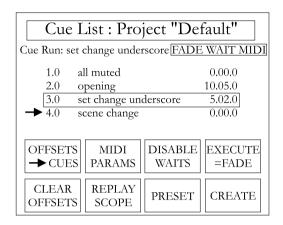
Playing Back Cues and Mixing

Cues can be recalled in sequence with the **[NEXT]** / **[LAST]** buttons, and non-sequentially by turning the Jog wheel, followed by the **[EXECUTE]** button. At the top of the Touchscreen is a "Last Cue" display, which will always show the name of the last cue recalled to the console from the automation system.



The Cursor, and Non-Sequential Cues

The cursor (inverted bar on the screen) will always highlight the last recalled cue, unless the jog wheel is turned, at which point the cursor acts as a "target" selector for the [EXECUTE] button, which will begin to flash along with the [CANCEL] switch as soon as the jog wheel is turned. When [EXECUTE] is pressed, the currently-highlighted cue will be recalled. If [CANCEL] is pressed, the cursor will return to the last recalled cue, and the [EXECUTE] switch will cease flashing.



The [NEXT] Predictor

This is a grand name for the little arrow (\rightarrow) which appears to the left of the cue which will be recalled by the [NEXT] switch. Even if the cursor is moved away from the current cue, since [NEXT] will always recall the cue sequentially following the last recalled cue, the arrow will not move, and can therefore be useful for location of the current scene position when scrolling through the list (remember also that the [CANCEL] switch will always return the cursor to the last recalled cue).

If [EXECUTE] is pressed with a non-sequential cue highlighted, however, then the arrow will jump to the cue following the cue just recalled out of sequence.

Mixing

The various control elements of the console (i.e. the faders, switches and rotary controls) may be moved at any time. They will always show the real value of the audio with the exception of VCA offsets to Input faders.

The automation data will be replayed to the console whenever a cue is triggered. Even if VCA faders are "scoped" out of the cue, the positions of the VCA Master Faders will be taken into account when applying the stored data to the console. In other words, if a VCA Master fader is at -20dB, the recalled gain values on ALL slave channels will be [the stored value] + [-20dB].

Isolate

There is every possibility that an input source might change drastically in nature due to an unforeseen circumstance - mic's slipping, people moving etc.. It is clear that the stored automation data will no longer be valid for those channels. The immediate solution to this problem lies in the [ISO](late) switch, which lies at the bottom of the central Encoder tray on the Input surface.

Pressing [ISO] on a channel will isolate the channel from the automation system. This means that the whole channel will effectively be in "manual" mode, and none of the stored changes will take effect. This remains the case until such time as [ISO] is deselected, after which any further cue selection will result in the automation values being written to the channel in question. A more advanced solution lies in offsets (see below).

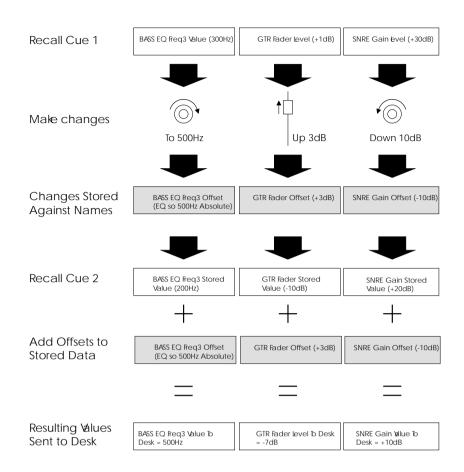
Offsets

Live sound is highly unpredictable. Actors, musicians, and even sometimes the audience can vary from night to night to such a degree that stored automation data can become at best inappropriate, and at worst completely wrong.

With a conventional analogue console, there is no problem. If a mic slips, for example, the operator of a traditional desk can simply make an adjustment to a gain, and that change will not be removed until another adjustment is made to the same control.

Unfortunately, this is not the case with an automated console. Taking the above example, the operator of Broadway could easily change a gain setting on the appropriate channel, and the sound would be corrected. As soon as the next cue was selected, however, the stored value for the gain on the affected channel (which, in all likelihood is the same as it was in the previous cue) will be recalled to the console, thus "undoing" the change which was so carefully made.

Broadway deals with this problem by using "Offset Groups". The idea is that, during run-time (i.e. whenever "Live" mode is selected - see "Arming and Saving Offsets" below), any changes made to the parameters on any channel will be stored as offsets to the stored data, rather than absolute values. The changes will be related to the name of the channel which was edited, and any further instances of that name from scene to scene will result in the appropriate offset being added to the stored parameters when the cue is recalled.



The only exception to the offset rule is in the EQ section - since it makes no sense to "offset" frequencies and widths, EQ parameters are treated as absolute values, which will supersede any stored data for that name. The console effectively "isolates" any EQ parameters which are manually adjusted, removing them from the automation. Remember that this isolation is unique to THAT PARAMETER on THAT CHANNEL for THAT NAME (offset group). It will not affect any other parameters or channels.

At the end of the cue list (i.e. at the end of the performance), the system will ask whether the user wishes to write the offset values permanently to disk (on a pername basis), or to lose them with power-down. The former is required when the changes are considered general improvements, the latter when the changes have been to counteract some kind of unusual problem or circumstance.

There is, however, a further complication to all of this. There is every possibility that a radio microphone will be used by more than one person during the course of the performance, and the offsets for one user may not be appropriate for the next. To counteract this, Broadway allows up to eight names per physical rack input. It is therefore possible to cater for up to eight users of a radio mic (each with their own offsets to stored data), or even eight different instances of the first user.

The latter case would be particularly useful if dramatic changes in costume changed the characteristics of the sound such that different input channel configurations were required for each costume. A mask, for example, or a large hat could have a serious effect on tonal characteristics. The user would not necessarily want input adjustments for one costume to affect other costumes later on.

The solution is to assign a new name for the same input for each different "style" required. BOB, BOBH and BOBM might cater for BOB on his own, Bob with a hat on, and Bob in a mask. Each name would automatically have its own offset group, and therefore any changes made would be associated with the correct settings.

The Offset system is much easier to use than it is to explain. To the uninitiated, the system is completely transparent - the fact that every input has a unique name by default means that all inputs have unique offset groups, and any changes made once the "LIVE" switch is pressed will be stored as offsets against the name, and used to affect upcoming automation data. Changes will appear to "stay" from scene to scene, thus rendering the console similar in operation to a conventional analogue desk.

Absolute and Offset Channels

Each name (remember, up to 8 per physical input are available) on the system may be defined as ABSOLUTE or OFFSET. Absolute channels will always be reset to their stored values when a scene is recalled, regardless of any changes made to them earlier in the performance. This can be useful for sources which are required to always be set absolutely, such as recorded effects, emergency PA settings or director's mic.

Arming Offsets

When LIVE mode is selected, any offsets will immediately be applied to the appropriate channels. When it is pressed again to deselect the mode, the channel parameters will "jump" to their stored values, without the current offsets.

It is therefore possible to use the [LIVE] switch as a toggle between absolute values of the last recalled cue, and the same data with offsets applied. This can be useful when checking the type and degree of changes which have been made to channel parameters before committing those offset values to disk permanently. When LIVE mode is active, all changes to automatable audio parameters will be

Saving and Clearing Offsets

regarded as offsets.

If desired, the offset values of each parameter may be written to disk. This will take the current value (= stored value + offset), and make it the stored value. At the same time, the offset will be cleared (because it is no longer needed).

When the performance or rehearsal has been run, and the required changes made to the audio, the user has two choices - either clear the offsets (if they were created to combat unusual circumstances that night, or were just not right...) or write the offset values to disk so that they will be recalled forever more.

Clearing Offsets

To clear the offsets, press CLEAR OFFSETS on the Cue List page. This will bring up a list of all available inputs. Each input may be selected or deselected as required by touching the input name on the screen. Alternatively, ALL channels may be selected simply by pressing the SELECT ALL switch.

Once the desired channels have been selected, pressing DONE will present a final confirmation screen. If confirmed, the Offsets will be cleared.

Saving Offsets

Saving the offsets is a similar process to that for Clearing the offsets, but it may be performed PER CUE as well as per channel.

An important thing to remember here - although there may be many names (offset groups) for each physical input, only one of those is active in any cue. So, when writing data to the hard disk for a selected offset group, it will check each selected cue, and if the selected offset group is NOT active in that scene, it will NOT write the data. So if the name "BOB" is only used in Cues 5 and 6, writing the offset group "BOB" for all cues will result in only Cues 5 and 6 being updated.

To start the process, press OFFSETS -> CUES on the Cue List page. A list of the available cues will then be presented. Select the cues that are required to be updated, then press OK.

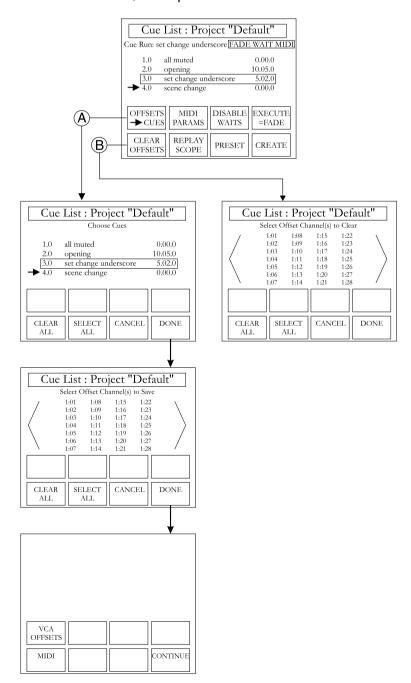
A list of ALL available offset groups is now shown. Remember that this will include ALL offsets from ALL channels (so there may be three or four entries for Input I, for example, if it has that many names).

As before, select the required channels, and press OK.

A final additional pair of options are now presented - VCA Offsets (offsets to VCA master faders - only worth doing if the VCA faders are being automated) and MIDI Continuous Controller Faders (where VCA Master faders have been assigned to act as MIDI Continuous Controller faders). Toggle these on or off as required, and press CONTINUE.

A final confirmation screen will be presented. If confirmed, the console will write the offset values for the selected channels to all selected cues.

Note that the offsets for any saved channels will be cleared at that point, as the stored value is now correct, and requires no offset.



Selectively Copying Data Between Cues

It might be useful to copy just part of a console setup - just the EQ, for example - from one cue to another. This is easily achieved via a slightly unusual use of the REPLAY SCOPE (see 6.6 Replay Scope) function, in combination with the fact that the console ALWAYS takes a snapshot of ALL data, regardless of the REPLAY SCOPE setting.

To take the above example:

- EXECUTE the cue which contains the desired EQ settings
- Scroll the cursor until it lies on the cue to which the EQ should be copied, and press [REPLAY SCOPE]
- In [CHANNEL], press [EQ] so that it is no longer hilighted, making sure that all other parameters are hilighted
- Recall that cue. The whole console will go to the settings in the new cue except the EQ (which is "scoped out"), and therefore left as it was in the last cue
- The desk setup is now as desired old EQ, different cue around it so
 move the cursor over the last recalled cue and press [UPDATE] (in MAKE
 CUE) to store the new settings in that location.

Channel Copy / Reset Channel

With the Channel Copy function in Broadway, it is possible to take the channel parameters from one channel and apply them to a number of different channels. There are two ways of achieving this:

- 1) Single channel copy
- 2) Multiple channel copy

I) Single channel copy:

Pressing COPY in the FUNC page of the ACS LCD will bring up the **COPY SIN-GLE I** page. The user is then required to tell the console the "source channel" - that is, the channel from which the data is to be copied.

The source channel is selected with a press of its associated **SEL** switch. This action will present the **COPY SINGLE 2** screen, which asks the user to select the destination channels to which the source channel data should be copied. Pressing **SEL** on any channel at this point will cause the source channel data to be copied to that channel. Once all copying has been completed, pressing **EXIT** (**F4** on the ACS LCD) will return the user to the **BASE PAGE** on the LCD.

2) Multiple channel copy

The **COPY MULTIPLE** LCD screen allows copying of channel parameters from a source channel to multiple adjacent (on the SURFACE) destination channels. The syntax for this page is:

- The user SELects the source channel (from which the data is to be copied);
 this selection moves the LCD to Copy Multiple 2.
- The second screen then asks the user to input the first of the destination channels. When this is SELected, the LCD will move to Copy Multiple 3.

■ The user is then required to input the last of the destination channels. The console will then copy the source channel parameters to the first and last SELected channel, and all channels WHICH APPEAR ON THE SURFACE between these two fader assignments. That is, if the first selection is fader I on BANK I, and the second selection is fader 3 on BANK 4, then the source channel data will be copied over ALL channels appearing ON THE SURFACE FADERS between those two locations (i.e. all faders in BANK I, all faders in BANK 2 and the first 3 faders in BANK 3)

As an additional aid to programming, when the SOURCE channel has been selected (in **COPY MULTIPLE I**), it is possible to reset the SOURCE to the default factory parameters for inputs. This is useful for resetting or "flattening" large sections of the desk with just a few button presses.

As with all Broadway functionality, there is no **UNDO** for the copy function. Remember, though, that the new channel information is not stored in a cue until a **[CREATE IT]** or **[UPDATE CUE]** is performed in the touchscreen. Should a copy be made in error, simply recall the original cue to return to the old desk setup.

GrAuxes and Matrices

The 32 GrAux sends and the 5 Main outputs in Broadway can all be routed to the optional Matrix outputs at user-set levels.

The routing from GrAuxes to Matrices is achieved on the Input Surface. The input surface has four modes, Input, GrAux, Matrix and VCA, selected from the three switches in the centre of the fader tray.

GrAux Mode

When none of these switches is selected, the surface is in Input mode. When GrAux is selected, the Faders will flip to show the first 20 (or, on the second Bank the last 12) GrAux master levels (replicating the controls on the Master surface). The 5 Main Outputs will also appear on the second Bank. The 6 rows of encoders above the faders will now show the send levels **from** each of the GrAuxes **to** the various available Matrix busses, with a single Matrix appearing on each row. The name of the Matrix buss will appear in the 4-character led matrix display in the centre of the encoder tray, and the up and down arrow switches will scroll through the possible Matrix outputs.

If [SEL] is pressed in this mode, the ACS will show the currently-selected GrAux. It will allow switching of the Insert, Mute and Solo switches, and level control of the various GrAux / Main to Matrix sends. GrAuxes may also be sent to the main outputs, using the OPI-5 switches on the ACS (this is obviously not available when Main Outputs are selected).

Matrix Mode

GrAux mode presumes that the user wishes to set the routing from audio groups to matrix outputs (usually speaker clusters) by group. If the user wishes to work by Matrix, they can select Matrix mode. This puts the Matrix master levels on the input surface faders, and the encoders above now show the contributions from each of the GrAuxes to the Matrix busses. The >and fl switches will scroll through all 32 GrAux sends and the 5 Main Outputs.

If [SEL] is pressed in this mode, the ACS will show the currently-selected Matrix, and will allow switching of the Insert, Mute and Solo switches, and, on the 16 rotary encoders, the level control of the various contributions from the GrAux busses.

GrAux and Matrix modes are effectively setting the same parameters, but the view is reversed. The user can choose whichever mode best suits the situation, and the data will be presented appropriately.

BROADWAY

8 Hints and Tips

Broadway may, at the simplest level, be operated as a regular, manual mixing console. The default project settings of Input 1-20 on BANK 1 of every surface, and 21-40 on BANK 2 of every surface allow the engineer to start mixing straight away, without reference to an operation manual or any training.

The easiest way to start mixing on Broadway is to select BANK I on input Surface I and BANK 2 on input surface 2. This shows the full 40 inputs of the system simultaneously, and allows immediate access to those channels.

Given the time to program a show, though, the Broadway system will deliver considerably more assistance to the operator. Fader assignments, channel naming, offset groups and VCA slave assignments will all help the operator concentrate on making the best-sounding and most consistent mix possible.

This document is intended to assist both the new and experienced Broadway operator in streamlining the operation of their desk, and in exploring new functions which have been added since the console's initial release.

Setting Up A Show

Since Broadway is an assignable and automated desk, there is a tendency to believe that "programming" means a long and complicated setup procedure. This need not be the case.

There are, of course, different levels to which an operator can program Broadway for a show. It is not necessary to complete all the tasks described below - any or all of the functions may be combined to suit the show to be run, and it is down to the user to decide how much benefit will be gained from each level of programming.

It is advisable, therefore, to decide IN ADVANCE of the programming roughly how the show should ideally appear on the console, and the "best guess" (given that the production may not even be in rehearsal yet!) as to the type and number of Inputs, GrAux outputs and Matrix outputs to be used. This way, the user will avoid unnecessary programming time, and end up with a show which is only as complicated as it needs to be, and the information presented is approximately correct from the first day of rehearsal.

There is no point, for example, in naming 40 channels of tie lines and all the outputs, if only 20 inputs and 2 outputs will be used in the production. The default names will be fine for any unexpected repatching which might be required, and the user will waste their time setting up all the channels which will never be required.

A Few Questions Which Might Be Useful As A Starting Point:

Would just channel naming be appropriate?

Do the outputs REALLY need to be named?

Will the VCAs be reassigned from Cue to Cue, or just remain static?

If the VCAs are being used heavily, is there any point in reassigning the Input Faders, as most of the time will be spent on the VCAs?

Channel Naming

Not only does Broadway allow input and output channels to be named, but it also gives the user the opportunity to give a single physical input or output up to 8 names, ONE of which will be selected as "current" in each Cue.

This might seem odd, but consider a couple of examples:

(a) A radio microphone for a member of a chorus in a theatrical production is given to three different actors during the evening. Despite the pack change, the physical input into which the radio mic is patched will NOT change. Simply go to the FADER ASSIGN page, press the NAME column for that channel, and name the first, second and third slots, or "offset groups", as appropriate for the actors to use that microphone.

It is now possible to select in each Cue one of the 3 names which were entered in the above process. Note that the audio parameters will not change (unless, of course, audio changes are programmed into the Cue List...!), but the name WILL change, assisting the user in identifying the actor in question, and in helping the Radio Mic Assistant in the event of confusion during rehearsal.

(b) A guitarist in the pit is using two different guitars, but only playing one at a time. Using two different names for the input, the operator may now display the two different functions of that channel (e.g. OVDE for overdriven electric, and CLEC for clean electric). The appropriate names may now be selected for each scene, keeping the user up to date with the instrument which the musician will (or should...!) be using.

Note that Offset groups may have unique VCA assignments, so, for the above example, OVDE may be in VCA 5, but if CLEC is selected, it will have its own VCA assignment, and will not inherit the VCA master of OVDE. VCAs are assigned PER OFFSET GROUP and NOT per input.

Fader Reassignment

The conventional method of Fader Assignment on Broadway (press INPUT in the Fader Assign page, which then asks for a Rack, input and Offset Group of the required input) can be a time-consuming process. There is a shortcut to this task which can greatly accelerate the process.

Pressing the FADER column (i.e. the 1:B1:01 entry) will bring up a list of all the inputs available, sorted numerically. This list will show one entry per channel, and will show the NAME of that channel if a name has been programmed. Since the console already knows the relationship between an input and its name, the user need only refer to channel NAMES from now on, and can forget about the "real" input location of channels in the rear of the racks.

So, if names have been assigned to all inputs, the task of assigning "Kick" (on input Rack I, input 2) is as simple as pressing the FADER column at the desired location, and pressing "Kick" on the list. The console will then make the necessary arrangements to place input 1:02 onto that fader location.

Sensible Arrangement

It may be perfectly acceptable for some applications to leave the default arrangement of inputs to surface faders for use in the show. However, sensible arrangements of faders can make troubleshooting and the mix operation more fluid and intuitive.

A Couple of Useful Ideas to Bear in Mind:

If the basic default setup is what has been used successfully until now, then leave BANKS I and 2 the same on the surfaces. On higher-numbered BANKS, try assigning the various components of the Orchestra (e.g. Rhythm, Percussion, Brass, Woodwinds) to a different BANK for each type. Leave at least one BANK (or more, depending on the number of radio channels) exclusively for RADIO MICs. This system allows faster access to certain types of instrument when mixing, so certain elements of the "sound" may be selected for viewing and adjustment. Of course, since BANKs I and 2 have not been changed, the user may still operate as before if required.

As a backup measure, try assigning BANKs I-4 as required on input surfaces I and 2, then assigning BANKs 5-8 on input surface I as per I-4 on i/s 2, and BANKs 5-8 on i/s 2 as per I-4 on i/s I. In the event of failure of one surface, this programming style gives an immediate replica of the 4 "lost" BANKs via the "redundant" surface. We refer to this type of surface programming as "reverse assignment", and it does give peace of mind to users who require 4 BANKs or fewer.

Ripple

Ripple is a system which allows the user to make changes to the whole show by editing just one Cue.

Note that NAMES (the 8 names per input and output) are changed per PROJECT, and NOT per CUE. So, on an unprogrammed system, changing the name of the default offset group (the first group) will effectively change the name of that input throughout the show.

However, fader assignments, VCA assignments, offset group selection and stereo channel selection will follow the Ripple law.

This works as follows:

When a change is made to an assignment of the type listed above (for example an input to fader assignment), the console will leave that change active on the surface until an incoming Cue makes a manual reassignment to the same fader. So, for example:

Cue I	Fader I is input 5**
Cue 2	Fader I is input 5
Cue 3	Fader I is input 5
Cue 4	Fader I is input 5
Cue 5	Fader I is input 5
Cue 6	Fader I is input 5

The six Cues shown above are set up in a show. The assignment was actually set up in Cue I, and this is therefore a "hard" assignment (**). Since no changes are made below, the console presumes that the following Cues require the same assignment.

The user now wants to assign input 7 to fader 1 in Cue 3. The result is:

Cue I	Fader I is input 5**
Cue 2	Fader I is input 5
Cue 3	Fader I is input 7**
Cue 4	Fader I is input 7
Cue 5	Fader I is input 7
Cue 6	Fader I is input 7

Cue 3 now assigns input 7 to Fader I, and this is a "hard" assignment. Because the assignments below were "soft" (presumed by the console, not manually set), they are replaced by the same information as Cue 3. If the user only wanted that change in one Cue (Cue 3), then it is now necessary to assign input 5 to Fader I in Cue 4. This results in:

Cue I	Fader I is input 5**
Cue 2	Fader I is input 5
Cue 3	Fader I is input 7**
Cue 4	Fader I is input 5**
Cue 5	Fader I is input 5
Cue 6	Fader I is input 5

The list now has 3 "hard" assignments, namely those in Cues 1, 3 and 4.

This principle is particularly useful in assigning VCA slaves across the whole desk AFTER a show has been programmed with many Cues. Simply set up the required VCA settings in the "top" Cue, and the console will not change those assignments until it receives a specific command to do so (via a new VCA assignment set).

Bear in mind that ripple also takes effect when a Cue is INSERTED into the list. It will affect all Cues below it until each parameter reaches a "hard" assignment.

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Scope

Scope is a very valuable tool in updating a show which has already been created, or in "rehearsing" certain parameters through a show without writing them permanently to disk.

Scope simply tells Broadway, on a PER CUE basis, to ignore certain types of data when recalling that Cue. Note that ALL data is stored when a snapshot is taken, and the user cannot change this rule, but what they CAN change is what is recalled.

For example, in a 10-Cue show, it might be desirable to try some different fader settings to those which have been stored in the Cues. Simply set up the desired fader positions, then go to REPLAY SCOPE on each Cue, and toggle FADER to OFF. The console will now ignore all stored settings for the faders from Cue to Cue, and will effectively leave them in "manual" mode.

Scope may also be used for UPDATING Cues which already exist. Using the above example, let us presume that the new positions of the faders are correct, and should be added to each Cue. Recalling the Cues in sequence will now present the desk state as per each Cue, but the faders, as before, will be manual and remain the same. So each Cue is recalled as before, but with the new fader settings. Now pressing UPDATE CUE fader each recall will store that new desired setup into the appropriate Cue location.

Problem-Solving

The DETECT NETWORK page on Broadway is a valuable tool in discovering which network units ("nodes") are active, and which are not. Should a problem arise with a unit (power loss, card damage, etc.) such that the device is no longer available to the system, then the user will be alerted via the Detect Network page, which will automatically appear when the user is on the CUE LIST or MAKE pages.

From the list of units, it is possible to see where the problem lies. Compare the "SYSTEM" column with the "PROJECT" column. The "PROJECT" column shows information about the project on the disk, including the number of inputs and outputs which the project EXPECTS TO FIND. The "SYSTEM" column shows the same information, but this time what the console ACTUALLY HAS. If all is well, the number of inputs and outputs should match up. If the input number is short, then the lnput Rack has caused a problem. If the outputs are short, then the output rack is the cause.

If the number of Input Surfaces does not match, then there is a problem with an Input Surface (note however that some users choose to change the number of input faders on their system after a project is created).

More Flexibility In The Hardware

There are a number of slightly unusual ways in which to use the rack hardware in Broadway to achieve even greater flexibility in any given system size.

Those users needing more inputs to the console, especially those requiring returns from FX units or audience playback devices which require only level control, can take advantage of the number of GrAux outputs on the console.

GrAuxes which are either unused, or acting simply as Aux Sends to external devices, can act as additional line inputs into the console. Simply feed the input signal into the Insert Return of the console, and switch the Insert point IN. This will result in the GrAux fader carrying the input signal, and the meter showing the post-fade level. The input signal may then be routed to any of the Main outputs, or the Matrix Outputs (if fitted).

Of course, the limitations here are that there is no Mic Gain or EQ on the input, and only Level, Mute and Solo control are available. This may prove to be too much of a limitation, but might help the user out of a problem when a large number of Line level sources are suddenly required to be mixed together.

The manner in which not only the unused GrAuxes but also the GrAuxes used as sends may be used for this purpose is by attaching the device to be addressed by the GrAux send to the Insert Send of the GrAux rather than the Output. Level control of the final send level is lost, but this may be a compromise which the operator is prepared to accept. Note that the Insert Send will always be active on the GrAux outputs, regardless of the status of the associated Insert IN switch.

This setup might also offer a simple way to set up a large number of Recording sends to tape, which are FIXED at line level, rather than potentially adjustable via the faders. The user would therefore be able to "exchange" 16 (for example) GrAux sends for 16 level-controlled Line inputs and 16 fixed-level tape sends. Given the large GrAux capability of Broadway, this tradeoff may make sense for those users using just a few sends and Groups.

BROADWAY

9

Software Upgrades

Software Upgrade Notes

In order to upgrade Broadway software, you will need a main system disk, containing the different type of software for each node, and the data files (DATA.DAT) appropriate to your size of system.

Note that, although version 1.106 and 1.2.16 used the same DATA.DAT files, Version 2.xx and above use NEW data files, which are different to ALL earlier versions. Version 2.xx MUST be loaded in conjunction with v2.xx DATA.DAT files to operate correctly.

Retrieve all the files in the v2.xx directory on the Website, and one of the DATA files in the DATA directory (appropriate to your system size), then follow the instructions below.

Installing a New Version of Software

Before commencing the upgrade, it is important that you have a Floppy copy of your existing software available in case of problems.

1) If you have a working system running older Broadway software:

Copy the '.HCA' files as appropriate for your system onto a blank formatted disk. You will need either an OUTIN_RK.HCA or OUT_RK.HCA dependent on system size.

If your system has 40,80,120 channels you need the OUT_RK.HCA file, otherwise you need "OUTIN RK.HCA"

Rename your DATA file (e.g. **60-40-0.gz**) to DATA.DAT, and copy this to the Floppy disk along with the other files (or onto another blank Floppy if the first is full).

You should end up with:

CONTSUFI.HCA MASTSURF.HCA AUDIO_RK.HCA

OUT_RK.HCA (or OUTIN_RK.HCA, as required)

VCAEXTN.HCA (if you have a VCA Extender Surface)

DATA.DAT (appropriate to your system size)

Insert this disk into the Broadway Master Surface, navigate to the Software Upgrade page, and press the "Load Floppy" button.

Once all files have been copied, press "Upgrade All Nodes"

After this has completed, press "Clear SRAM".

You should now reboot the whole system. See below for the boot procedure.

2) From a completely blank Broadway system:

Copy the individual '.HCA' files onto individual disks, renaming them to have the extension '.GZ. Thus OUTIN_RK.HCA becomes OUTIN RK.GZ

Copy the file LOADER.ABS onto each floppy disk.

The disk containing OUTIN_RK.GZ or OUT_RK.GZ should also contain the file DATA.DAT, FILES.LST and FILES.DAT.

In order to load software onto the CPU cards in audio racks or input surfaces, you will have to plug those CPU cards into the master surface.

For each CPU card that you plug in:

- 1) Turn off the master surface
- 2) Plug in new CPU card
- 3) Place floppy disk created above for that rack/surface into drive
- 4) Turn on the master surface
- 5) The floppy light should come on for a minute or so, while it loads the software
- 6) Once floppy light has gone out, remove floppy disk
- 7) Repeat for each CPU card

Boot Procedure

After the installation of new software and the subsequent SRAM CLEAR, the software must be started carefully in order for it to start up correctly. After new software, the input racks must be started in the correct order i.e. input rack for channels I-40 before the input rack for channels 4I-60 (OUTIN_RK). The output rack should be the LAST rack turned on.

Please wait at least 20 seconds between turning on each rack or control surface.

Once your system has been started once and is fully working, you may start the console with a single action (system power). The boot order is only significant when the SRAM has been cleared.

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Surface Fader Calibration

The console supports localised Surface fader calibration on booting, by holding a number of switches when the Surface is powered up.

To perform a Recalibration, select the correct Surface type and fader area from the list below, and reboot the Surface holding the appropriate SOLO / FUNC switches:

Control surface Channel I Solo + Channel 4 Solo	:- calibrates CH I-10
Control surface Channel 11 Solo + Channel 14 Solo	:- calibrates CH I I -20
Control surface Channel 1,2,3 & 4 Solo	:- calibrates CH I-20
Master surface VCA fader Solo + VCA fader 4 solo	:- calibrates VCA 1-8
Master surface GrAux FUNC + GrAux 4 FUNC	:- calibrates GrAux I-4
Master surface GrAux 5 FUNC + GrAux 8 FUNC	:- calibrates GrAux 5-8
Master surface GrAux 9 FUNC + GrAux 12 FUNC	:- calibrates GrAux 9-12
Master surface GrAux 13 FUNC + GrAux 16 FUNC	:- calibrates GrAux 13-16
Master surface Output FUNC + O/P 4/5 FUNC	:- calibrates O/P I-5
Master surface GrAux I 2,3 & 4 FUNC	:- calibrates GrAux I-12 & O/P I-5
Master surface VCA I 2,3 & 4 SOLO	:- calibrates VCA I-8 GrAux I-12 & O/P I-5
VCA extender VCA fader I solo + VCA fader 4 solo	:- calibrates VCA 1-12

Once the fader recalibration is completed, the Surface will continue to boot as usual.

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Appendix

A

The Harman Communications Architecture (HCA) -An Overview

Introduction:

Why Is It Necessary To Develop a Control Architecture?

In the last few years, there has been a considerable increase in the number of parameter-based, software-controlled professional audio units. As the benefits of such flexibility and configurability become clear, each manufacturer will further develop their own proprietary feature set, parameter list, and sometimes even their own data structure to suit their particular application.

Whilst this explosion of ideas and feature sets offers clear advantages to the end user in terms of the configurability of any given unit, it is clear that, with the arrival of each new piece of "black box" audio hardware, the chances of any two connected machines being "compatible" (i.e. able to exchange information in one form or another) are substantially reduced. To restrict manufacturers to a basic feature set and parameter list would certainly go a long way towards reducing these incompatibilities, but would also inhibit the creative design process and evolution of new hardware.

The alternative to such restrictions is to develop a protocol for intercommunication of various devices, which would allow any interconnected machines from compliant manufacturers to edit and share each other's parameter information. Whilst this network intercommunication protocol is required to be sufficiently generic to allow communication with any connected device, it must also allow manufacturers to retain design autonomy within their own equipment. As far as the user is concerned, it is easy to see how such a co-ordinated and highly-compatible control interconnection protocol would be invaluable in system setup and operation. Not only would the creativity of the user be concentrated on the performance aspect rather than the technical details, but new methods of using controlled hardware could lead to innovative audio mixing.

It is just such a realisation which has been circulating through the audio industry for quite some time. The development of the Harman Communications Architecture has taken place in direct response to just this perceived need in the Professional Sound Reinforcement market for an "intelligent" communication protocol between networked devices.

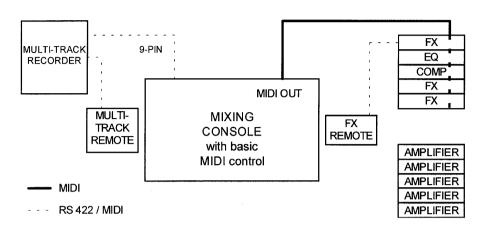
The abilities of any new protocol were required to extend beyond the single-ended "blind" remote control systems (where the message is sent with almost no knowledge of the target item) with which today's pro audio system designers are forced to work, offering instead a managed and co-ordinated approach to system integration.

A considerable amount of work in this area has already been completed over a number of years by the AES SC-10 committee, the culmination of which was a proposal document for a new protocol - the AES-24 standard. Although many aspects of HCA closely mirror in principle the proposals made in the AES-24 working document, there are some areas in which the HCA development team felt it necessary to expand upon and, in some cases, revise some of the ideas therein.

What Is a Control System?

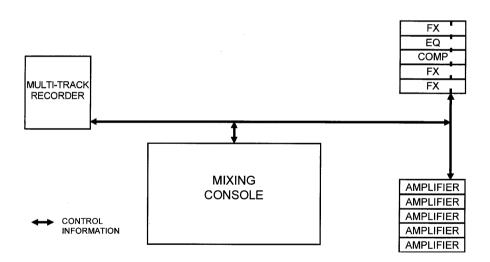
Let us first consider a typical conventional system for multitrack recording:

Fig. 1. A Conventional "Integrated" System (Audio Not Shown)



A control network in this context is any intercommunication protocol which allows the controlling elements of a unit to be remotely accessed and updated.

Fig. 2. An Example Of A Basic Control Network (Audio Not Shown)



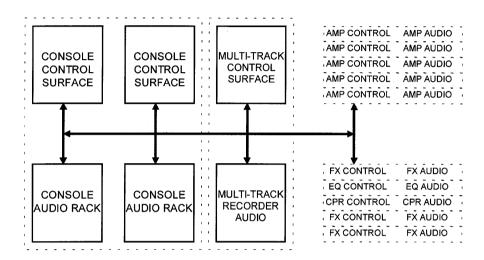
The ideal control system would allow the storage of parameter data for later recall, and would allow the user to access any of the "audio" components from any of the "control" components. This requires that a list of the various control and audio "destination" resources available on the system be drafted and stored by the controlling system. To achieve this, a fundamental distinction needs to be made between the Control part of any given unit, and the Audio part.

Enter HCA

How does HCA achieve these goals? HCA is a system control and management protocol, which has been designed to facilitate the interconnection of networked audio devices. HCA considers all components within the system - faders, eq, rotary knobs, DSP - as separate elements, even if some of these reside within the same unit. By default, the control elements on the front panel of a reverb will obviously access the parameters within that reverb, but they may now be "repatched" to access any other element in the system.

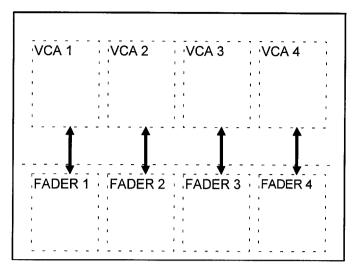
This may better be explained by way of an updated version of Figure 2:

Fig. 3. Control Separated From The Audio



The most important point to note about the system in Figure 3 is that in all cases the control elements have now been separated from the audio, although they may still be within the same unit. This point is vital to the principle of HCA. All of the HCA component parts are "virtual" components, that is, they may live in software on any device within the network. Thus the controlling component of an Eq device, for example, might be hardware controls on the front (which would send software information to the Eq audio components) or a "glass screen" virtual software interface, on a PC, touchscreen, LCD, or any other interface device. The hardware / software nature of the elements of the system are of no particular importance to HCA, just as long as the system knows how the element should be controlled, and what control capabilities it itself has.

If we now consider the various elements within the control and audio components of each unit on the system, such as faders and rotary knobs for the control sections, and VCA and Eq components for the audio sections, it is possible to build up a list of all the individual components available within the system. By default, it is clear that most control items will have a pre-determined target audio function, such as:



Control	Mapped To Taget
Fader I	VCA I
Fader 2	VCA 2
Fader 3	VCA 3
Fader 4	VCA 4
etc	etc

Since the system now has access to all of the components in the system individually, it is a relatively simple matter to set up a routing matrix similar to the above table to route any control object to any destination audio object - the beginnings of a very powerful control system.

Note that the interconnecting cable is not defined. It is important to appreciate that no particular transport medium (I) has been defined as part of HCA - that is, HCA is a Networking and Data Transport system which uses some defined primitives (2), and is not based on any particular computer format or connector type. The only restriction is that the connecting medium be as fast as each unit requires. This principle carries with it two important advantages -

- Any new, faster transport media which arise may be easily adopted and integrated.
- The transport medium for any device need only be as fast as the function of that unit requires, with a cost to match. It would not be sensible, as an example, to require that a simple guitar pedal be fitted with the same interface as a fully-assignable audio mixing console.

- A transport medium describes the manner in which information is physically transferred from unit to unit within a network. Ethernet, MIDI and C3 are all examples of transport media.
- 2 A "prinitive" is a command called at a high level by a controlling system, but executed via a series of proprietary functions by the carrying operating system. HCA is not interested in the resulting code used, only that the data transport functions which HCA requested with the primitive are executed correctly, and that an appropriate "tally", or answer, message is received in return.

System Overview

First of all, the system needs to know the location (on the network) and function (whether a control or audio function, parameter ranges etc.) of each of the components in the system.

Objects

HCA translates the individual proprietary audio hardware and, where appropriate, the front-end software components of each networked device into a series of "objects". These are the most basic element in the HCA world. Every software parameter, every audio component, every hardware control is reduced to the status of an "object". Fundamentally, there are just two types of object - a "generic" object and a "non-generic" object. A "generic" object is one which has similar functionality from unit to unit, regardless of manufacturer - e.g. fader, encoder, VCA, etc.. A "non-generic" object is unique to a manufacturer or a particular unit. This type of object must announce its presence in more detail - a process which will be discussed in more detail in the section Object Discovery below.

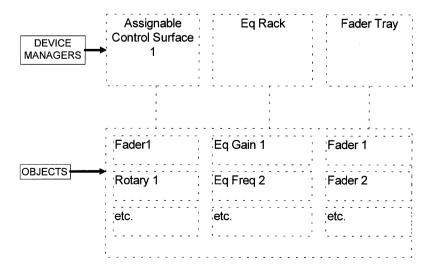
An object's type name is derived from two categories - manufacturer and object type. Thus generic/fader (i.e. a control surface object) or lexicon/delay_time (an audio object) are possible examples.

Device Managers

Taking the example of an assignable console, it is clear that there will be a vast number of faders, rotaries, audio components etc., all of which comprise the one "console" unit. Dealing with each of these components individually would require a larger bandwidth (data rate). This problem is solved by the Device Managers. These are special instances of an object, which can create groups of other objects. So a device manager might create a "control surface" object which consists of several encoder objects, fader objects and switch objects. It might equally produce a "processing" object which comprises several Eq objects, and some Limiter objects.

These may then be handled as single entities by the HCA architecture, although all of their individual addresses on the network are retained relative to the device manager. It is easier to think of these managers as similar in function to sub-directories on a computer disk. Take the example of a 40-fader, 20-encoder control surface. The device manager will allocate a position ID to the new collection of objects, and names it "Assignable Control Surface". Now, instead of searching individually for 40 faders and 20 encoders, the system simply looks for "Assignable Control Surface I" (which has been pre-defined as 40 faders + 20 encoders).

Fig. 5. Device Managers "Group" Objects to Assist in System Building and Communication



The Registry

A major goal of the HCA system was to achieve as close to true "plug and play" hardware control capabilities as currently possible, and to eradicate as many of the integration headaches traditionally associated with attempts to remotely control and access information within interconnected devices, be they Dynamic Processors or Multi-Track Tape Recorders. Given the rapid development of new devices and working methods, any new protocol has to be relevant to today's market, whilst remaining sufficiently open-ended to accept new equipment and different working practices. Figure 2 could operate quite well as a control system if all of the equipment knew exactly what the other machines were, and how to control them. So how does HCA achieve this?

HCA is an "intelligent" system; that is, it will detect and integrate any new unit connected to the system, and determine the type of control / slave operation of which that unit is capable. The location and function of every component of the system is stored in an area called the registry. This area may reside within any of the units on the network, and stores information on all the objects available to the system - the HCA equivalent of an office cardfile system with contact information. The registry operates as a database, or local "address" book, with information about all objects available to the system. The information includes the physical "address" of the object (i.e. the network location at which the object resides), the type of object, and the parameters required to control it. There is only one registry per network domain (3), which means that the "administration" is dealt with by one central processing unit, which brings with it certain advantages.

Object Discovery

When a new device is connected to the system, its own device manager already knows the resources available to it locally, so it now needs to tell the network about the objects it has to offer, and what information it requires in return (4). The device manager first checks for the presence of a registry. If none is forthcoming, the new device will become the registry, otherwise it will begin negotiations with the registry to inform it of the objects within the new device.

The registry now knows the location of the new objects (sub-grouped under the device manager name), and adds them to its database. Now any control unit wishing to find one of the new components simply asks the registry where that unit resides, and how it should control the target object. This saves the task of searching the entire network for the desired unit, which would be both time-consuming and wasteful of network data flow (bandwidth).

One possible disadvantage of such centralisation of information is that a registry failure could be catastrophic. The system therefore carries a "mirror" (copy) of the register in another unit on the network for security backup.

- 3 A network domain is an area of the HCA network in which all of the units are in permanent contact with each other from power-up.
- 4 If the unit has no battery-backed memory, it may load its HCA configuration data from another machine on the network-the so-called "bootstrap server". The unit requesting information is said to be the "bootstrap client".

Handles - Dynamic Position ID's

Within any network, the problem of unit addresses is always a major issue. It would be easiest to presume that every unit had a unique address, and would always be referred to at that address. Unfortunately, this would require very long addresses, and some fairly tight regulation on the part of manufacturers.

The alternative to the static ID solution is to allow the system to allocate new "soft" ID's to every component on the system each time the network is powered up. The disadvantage with that idea is that the network configuration depends rather upon knowing "where everything is", and "re-naming" time could be very long.

The HCA solution is something of a hybrid of these two concepts. Each object has an ID, derived from the manufacturer's name and the object type and the "father" device manager's chosen position ID for that particular object. There is no guarantee that this will be unique, so the system can update this address if necessary. When it comes to "runtime", i.e. when the network begins to operate, the registry allocates every object on its database a short, unique (32-bit) address - a "handle". This handle list is a sort of localised table of contents recompiled each time the system is reconfigured, with each entry in the list being given a runtime ID, which may be different every time the system is run.

For example, if three objects existed on the network - soundcraft_fader_I, soundcraft_fader_2 and soundcraft_VCA 2, they could be allocated handles thus:

Handle	Object Name
1	soundcraft_fader_I
2	soundcraft_fader_2
3	soundcraft_VCA_2
etc	etc

These are obviously not the true data structures, but the principle is the same.

The System Builder

We now have a detailed library of uniquely addressable objects, with all of the information necessary to use them within an integrated system. We now need some way in which to configure the layout of the system, and to make the patches, or "links", between the control components and the audio components.

This task is achieved by the system builder. This object allows all of the components to be "interconnected" (in the virtual space of the network), and given parameter ranges, labels etc.... From the user's point of view, the system builder will almost always be implemented in the form of a glass-screen (PC) front-end, but the same guiding principle applies - that the actual system builder area of HCA may reside within any device on the network.

The combination of the System Builder and the Registry (along with the latter's ability to detect new devices upon power-up) is what makes HCA so flexible and so powerful. The centralisation of the object directory inside the registry means that complete network scanning (or "polling") should only very rarely be necessary. This means more space for other, more important messages, such as parameter change and, perhaps one day, the transmission of digital audio or video.

Flexibility and Invisibility

HCA is very flexible, and will configure itself to suit the system setup - none of the software within HCA is required to live in any particular place throughout its existence. Every time the network is powered up, the Registry and System Builder will check the status quo: if all is well, the system simply sets up the links as per the last saved session; if there is a problem (i.e. if a device has changed, been added, or even been removed), the software will perform the tasks necessary to make the new network function correctly. In that sense, HCA is an "invisible" system - the power of the software means that the headache has been taken out of system configuration and remote parameter access, and the user is left with devices which will either just get on with the job, or which can easily, if necessary, be reconfigured to perform another task.

Conclusion

The HCA system as a whole offers a level of control interoperability which has long been needed given the increasingly complex nature of the hardware / software combinations which make up the majority of new devices on the market.

Operationally, HCA is essentially a transparent system. The user only has to decide the configuration in which the system is to be used, and the software will do the rest.

As with any protocol, HCA depends upon the co-operation of a number of manufacturers if true intercompatibility is ever to be achieved. HCA has been designed to be as flexible as possible, and as generic as possible. It offers a simple and effective series of basic commands via which any unit can communicate with any other, whilst, thanks to the device managers, providing the freedom for manufacturers to build an appropriate control system unique their own device.

Manufacturers and software developers have not been slow to appreciate that the audio industry has long needed a generally-accepted control information and system management medium, and HCA is certainly equipped to be a front-runner in this race. The intercommunication protocol will be freely available to any interested parties, and the level of integration from unit to unit is therefore in the hands of each individual manufacturer - and, as such, HCA will only be as powerful as the support it receives from the industry.