"AXION"

USER MANUAL

AXION Owners Manual

CONTENTS:

Dear client,

Thank you for selecting the D&R Axion series.

The Axion was created using the latest in computer aided design and assembling technology and incorporates the most advanced circuit components which results in the Axion being another D&R product unsurpassed in the electronics industry.

We are confident that you will be using the Axion for many years and wish you much success.

We always value suggestions from our clients and we would be grateful if you could complete and return the questionaire included at the back of this manual once you become familiar with your Axion. We learn from your comments and appreciate your time.

With kind regards,

D. de Rijk
President, D&R Electronica b.v.

AXION soundreinforcement Console

The D&R Axion series is a balanced, 8 buss Front Of House (FOH) soundreinforcement mixing console designed to take the central role in a life performance facility.

The Axion is completely modular and can be configured to precisely suit your particular system requirements.
To become completely familiar with your Axion and gain the maximum benefit from its use, we recommend that you read this manual thoroughly. It will provide important information about all aspects of the Axion including; installation, operation, and servicing.

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THE CHASSIS SYSTEM

1.0 The Chassis System

The Axion is available in two frame sizes, accepting 44 and 60 input modules. The basic frame includes five blank modules, two are located on the extreme left and right of the frame and two more are on the left and right of the master section, and there is one more right from the group matrix modules. The extreme left and right blanks cannot be used for input modules as they conceal mechanical constructions, wiring, and power distribution, as is the case with the blank right of the mastersection. However the blank on the left side of the master section and the right side of the group matrix modules can be replaced with input modules if visual seperation of master and group modules is not requested, in this case the frames can accept 46 and 62 modules.

The frame 44 will fit 44 input modules (32 input and 12 stereo modules), 6 master modules, and 8 group/matrix modules.
The standard configuation has (from left to right) 1 blind module, 32 input modules, 1 blind module, 6 master modules, 1 blind module, 8 group/matrix modules, 1 blind module, 12 stereo modules and 1 blind module. Custom configurations are available at no extra charge.

The frame 60 will fit 60 input modules (48 input and 12 stereo modules), 6 master modules, and 8 group/matrix modules.
The standard configuation has (from left to right) 1 blind module, 24 input modules, 1 blind module, 6 master modules, 1 blind module, 8 group/matrix modules, 1 blind module, 24 input modules, 12 stereo input modules, 1 blind module. Custom configurations are available at no extra charge.

Both the master section and group/matrix modules can be installed wherever most suitable, but the request must be made at the time of ordering.

THE MASTER MODULES

The Axion has six master modules which are completely modular. All inputs and outputs are located on the back of the meter bridge. The paragraphs below give a description of each module section.

2.1 Solo section

The Solo section has a master PFL volume control with a center detent for nominal levels and a master AFL volume control with a centre detent and a Channel to Solo In Place switch with a switchcover to prevent from accidental switching during life performances. When this switch is in the up position, all solo switches on the input modules are in the "PFL" mode when activated. A LED indicator is also fitted next to the PFL level control to show when a solo circuit is activated. Both AFL and PFL controls have a range of + and - 20 dB.

2.2 The MONITOR section

The Monitor section contains the electronics for monitoring all signal paths in the Axion.

Monitor Source switching

From the top of this section, there are the five Monitor signal sources. With all these switches in the up position, the Monitor will not receive any input signal. This has been designed to be able to listen to any combination of input source switching as is desired during performances.
The first switch activates the Listen microphone fed from a seperate mic input at the back of the console. This mic input can be used for checking sound in the auditorium of a theatre where seperate control rooms are used for the mixing console. The listen Mic input can be phantom powered by jumper settings on the printed circuit board.
The second and third switch make it possible to listen to 2 track inputs from CD players.
The fourth switch lets you check the Mono output of the console and the fifth switch checks out the stereo main outputs, post main faders.
The last switch lets you monitor pre or post the main mono/stereo faders. Now it is possible to check the main mix without sending signal to the speaker systems. All selected input sources can be summed.

The monitor control has a seperate output for nearfield monitoring. The Phones control has its own outputs on the frontpanel in a recessed section close to the faders. A mute switch cuts the very powerfull headphones amps.

The main stereo faders are mono audiotaper faders and control the overall outgoing level coming from the main stereo mix bussamps. A 10 dB gain is available at the output, which is set to be +4dBu nominal.

Monitor level

The Monitor level controls the total outgoing level to the monitor. When in the full clockwise position, a balanced signal of +4 dBu is given out to the monitor amps. It is important to have the monitor amps correctly adjusted. You should adjust the monitor amps (input level controls) to an undistorted level with the Monitor level control fully clockwise.

NOTE: This alignment is imperative in order to avoid damage to the speakers, or in some cases, damage to the ears of the listener.

Two monitor Systems

The Axion has two monitor systems intended for use with headphones and nearfield monitors. We advise that nearfield monitors be wired to the Monitor output.

Insert stereo

The insert switch switches the signal processors returns into the main signal path for both left and right outputs.

2.3 COMMUNICATION SECTION

The communication section is a very comprehensive and important part of the Axion console. It is vital in life performances to be able to communicate at any point during the show.
Inside the master section of the Axion there is a “Clearcom” “ASL” compatible master station to feed “Beltpack” systems of either brand, making communication easy.
The Talkback microphone (which has an input on the frontpanel as well as on the connector panel at the back of the console) can be phantom powered by jumper settings on the board.
The Talkback mic can be assigned to the following sections of the console individually or to all at the same time.
Direct output at the back of the console.
Aux output amps (if the aux enable switches are active)
Matrix outputs (if the aux enable switches are active)
Group outputs (if the group enable switches are active)
Mono output
Stereo output

Communications to phones

This switch brings the communication system into the phones to be able to listen to stage engineers calling you through their “beltpacks”

Phones to communications

This switch allows you to pass the monitor signal to the beltpacks. Stage engineers can pre fade listen to channels and have the same acces to all relevant signal in/outputs in the console as the main balance engineer.

Side tone

These trimmers allow you to reduce the send signal in the communication systems to be lower than the receive signal. This needs to be adjusted once depending upon how many “Beltpacks” are connected to the system.

Comm

This switch assigns the talkback mic signal to the communication system only.

Call “A” / “B”

These switches activate the optical call system with a flashing light on all connected Beltpacks when hit once, after holding down the call buttons a few seconds a sirene will start drawing attention when the “called” station has not answered.

Insert mono

This switch switches the return of a connected signal processor into the mono signal path.

2.4 RECORD OUT / ALT P.A.

This section provides you with a second stereo output from the Axion's master section. It can be fed from the either the stero output or the mono output. Both these signals can be taken pre or post the main faders. An AFL switch checks out the signal strength and sourcing.

2.5 TWO TRACK REPLAY.

The Axion has a seperate two track replay input with lineair fader and a two band equalizer.
Both 2 track A and B can be summed if necessary and send to the mono and or stereo main outputs. The stereo fader sends the 2 track signals post main faders directly into the stereo and/or mono output amps. Smooth fading in and out of 2 track and /or main mix buss sound is easily achieved.

2.6 OSCILLATOR SECTION

A sweep oscillator is fitted. The sweep frequencies are; 20Hz to 2kHz, and 200Hz to 20kHz. The level ranges from -35 dB to +20 dB with a detented mid-position of +4 dBu. The oscillator can be switched to pink noise to be able to check out frequency response of the auditorium.
The oscillator can be routed to the **aux mix busses, group busses, mono output, left output only, right output only,** and to a **direct output.**

NOTE: The monitor will **dim** 20dB when the oscillator is active. The meters on the Axion are peak reading meters and therefore read +4 dB when a sine wave with a +4 dB output level is sent to the meter. Measuring the +4 dBu output level of the channel or master with a AC voltmeter would give a 1.22 volt reading.

2.7 MUTE CPU SECTION

The Axion has the possibility to store/program mute settings in the channels to a maximum of 64 settings.
The RESET switch resets the CPU to a start position without erasing mute settings in the channels.
The SAFE switch prevents the computer from being activated.
The MIDI switch changes the display from displaying patches into midi channels

NOTE: THE MIDI SOFTWARE IS NOT YET IMPLEMENTED, BUT WILL BE SOON BY A "ROM" UPDATE.

The CHANNEL/PATCH switch lets you change the display from showing the midichannel numbers into showing the patch numbers used for storing mute patches.
The display serves different modes which will be explained later.
The STORE switch lets you store mute settings from the channels against a patch number.
The UP/DOWN switches let you step up and down through the 64 available patchnumbers.
The PREVIEW switch shows the stored mute setting under the displayed number by flashing the channel mute lamps, without changing the actual mute settings.
The RECALL switch activates the mute setting stored under the actual display number showed at that moment.
The A to H switches can be used to store 8 settings out of the 64 patches to be able to instantly recall a stored setting.

To use the Mute CPU it is advisable to follow the next instructions.

When the Mute computer is "empty" (does not have programmed mute settings) it is advisable to set the disply to 00 (max 63) by hitting the up or down switches. Longer pushing of the up/down switches will cause the display to run faster after a while.
Now program the necessary mutes in the channels and hit the STORE switch, which will store this setting under patch number 00. As soon as this patch is stored the display will show patch 01 which will be your next patch available for storing new mute settings from the channels. This sequence can be repeated up to setting 63 to achieve a maximum of 64 different patches.

To RECALL patches simply choose the patchnumber by the up/down switches and hit the recall switch to activate that patchnumber. The recall switch will be lit now. As soon as you change the patch settings to another patch in the display the RECALL button lamp will turn off showing you that the display setting is not the actual channel mute setting.
The Preview function now shows the new display mute setting by flashing the related mutes in the channel. *THIS WILL NOT CHANGE MUTE SETTINGS!*

To store a specific mute setting under one of the 8 large recall switches A to H, simply hit the store switch and assign this setting by holding the store switch down followed by pushing the requested A to H recall switch.

2.8 AUXILLIARY MASTER SECTION

All auxilliary master sections are identical in function. All sections have level control, a talkback enable switch, an AFL switch, a Mute switch and a global pre/post switch.
Each Aux master controls the busses from the channels. The solo switch sends the aux buss signal to the monitors which is a post-fader signal. The associated LED lights indicating the activated solo switch.
The Aux mute does not mute the signal sent to the AFL buss, but mutes the outgoing signal to the aux outputs only.
The global pre/post switching in the Axion console is a convenient way of globally switching the pre/post feed of one aux buss for the entire console.
Aux 1 / 2 and 3 / 4 are switched globally in pairs. Aux 5 to 12 are switched individually pre/post per aux bus.

2.9 VCA GRANDMASTER SECTION

The grand master fader controls all assigned group VCA masters with an extra gain of 10dB over the group VCA master.

Master Metering

The Axion master is fitted with peak reading, high resolution, LED-bar meters with attack and release times which conform to world standards. The attack is 10msec. for a 20 dB range and the release is 1.5msec.The peak reading ledbars will show 50 dB of dynamic range from -30 to +20dB. This enormeous range precisely tells you what the actual headroom is that is available at any time.
The Axion has seperate meters for PFL and stereo AFL as well as stereo monitor.
The main left/right outputs are displayed on VU type meters to give an average level. The mono output is diplayed in VU on a led-bar, and the PFL/AFL and monitor signals are displayed on peakreading ledbars. The actual level displayed on the peakreading ledbars is the internal level of the console at that specific point. There is no 6dB down adjustment on the ledbars in the Axion. We did not want to create any doughts about the available headroom in the console.

If analog meters are mounted, when reading "0" on the VU meter you should have a +4 dBu or 1.22 volts on a volt / ohm meter.

THE INPUT MODULE

3.0 THE INPUT MODULE DESCRIPTION

The Axion input module is a basic input design whereby all signal flow takes place from the microphone to the main outputs. Each input channel is shipped with a 25 segment LED bargraph meter which is a peak reading device with attack and release times in conformance with world standards. It reads the level at four points in the channel. At the pre insert point, at the post insert point, at the post EQ point, and at the post VCA point. The first LED in the bargraph is a power supply indicator. The following sections explain the many functions and features of the input channel.

3.1 CHANNEL ASSIGN SECTION

The channel assign switches are .located at the top of the module and send the signal to the 8 groups individually or to any combination of this. Assigment to the left/right busses and to a seperate mono output is possible.

The pan-pot can be inserted between the odd and even groups when necessary. The pan-pot is always active on the Left/right buses

3.2 THE INPUT SECTION

The input section controls all incoming signals from microphone and line inputs.

The GROUND-LIFT switch lifts pin1 from the XLR type mic input connector from ground.

A **+48V** phantom power switch for condenser microphones or direct boxes can be switched in or out of the circuit.

NOTE: IT IS ADVISABLE TO HAVE THE CHANNEL MUTED WHEN SWITCHING IN OR OUT THE PHANTOM POWER!

The ground lift switch is automatically disabled when the phantom power is switched on.
The ground lift LED will turn off accordingly.

Line switches the microphone input to line input on the channel. The line input has its own balanced input amp and is controlled by the active (dual) gain control.
The **GAIN** control is the single most important control on the console. With this control accurately set, it is possible to achieve the very best signal to noise ratio and maximum headroom required for high quality life sound. This control is for adjusting the **line / mic** input and has seperate electronics although only one knob adjusts the dual pot.

Phase is used to reverse the phase of any mike / line input coming from a mike or signal that may be out of phase with other mikes or signals. A successful method of checking for "out of phase" signals is to pan both signals to the centre and llisten closely to the mix. If an unexpected sound is heard or if something appears to be missing from the mix, depress the phase switches for those channels suspected to be in error. If the sound improves, then that channel was out of phase with the others.

If using multiple mikes on the same signal, such as drums, vocals, horns, strings etc., it is possible to create an acoustical phase cancellation. In most cases, physically moving the mikes a few inches will correct this phase cancellation.

3.3 THE EQUALIZER SECTION

Just ahead the equalizer section there is a variable 12 dB VCVS high pass filter with a Butterworth curve ranging from 20 Hz to 1kHz switchable in or out of the circuit independant of equalizer on/off switching.

The four-band parametric equalizer is unique in its design. There are four bands, the high and low are sweepable frequency with shelving characteristics with a boost or cut of 16 dB and the two mid bands each sweepable with a boost or cut of 16 dB. The HMF (High / Mid Frequency) and LMF (Low / Mid Frequency) can be switched to a narrower bandwidth from 1/3 to 2 octaves.

The **HF** (high frequency) section is a variable frequency shelving type sweepable from 2,000 Hz to 20,000 Hz with a maximum boost or cut of 16 dB.

The **LF** (low frequency) section is a varible frequency shelving type sweepable from 20 Hz to 500 Hz with a maximum boost or cut of 16 dB.

The **HMF** (High / Mid Freq.) section has level and frequency controls and is a constant Q type, therefore the bandwidth setting will match that of the level control. The frequency ranges from 500 Hz to 10,000 Hz and has a maximum boost or cut of 16 dB. The bandwidth is switchable between 1/3 and 2 octaves.

The **LMF** (Low / Mid Freq.) section has level and frequency controls and is also a constant Q type, therefore the bandwidth setting will match that of the level control. The frequency ranges from 50 Hz to 1000 Hz and has a maximum boost or cut of 16 dB. The bandwidth is switchable between 1/3 and 2 octaves.

All level controls are center detented making neutral positions easy to establish.

All frequency ranges have been carefully selected following extensive examination of all types of music which makes this equalizer a pleasure to work with. Noise and distortion are kept to an absolute minimum.

3.4 AUXILLIARY SEND SECTION

The Axion has twelve auxilliary send busses. **Aux**illiary sends **1&2** and **3&4** are on dual concentric controls. The top control is the send control for aux 1 and 3 and the bottom control is the send for aux 2 and 4. These Aux busses are normally used for stereo headphone sends. All four can be fed from either pre or post the channel fader by the global pre/post switching in the master aux sections.
Aux 5 to 12 have individual global pre/post switching per buss located in the master section.
All aux sends have individual mutes per send except for aux 1/2 and 3/4.
Aux 12 has a direct switch redirecting the aux buss 12 signal to a dedicated output on the back of the console. This output can be used for life tracking of individual channels, or as an aux send for a specific channel.

3.5 THE SOLO MUTE SECTION

The **SOLO** switch has two modes, pfl (pre fade listen) or a "destructive" stereo **S**olo-**In**-**P**lace system. Master status switching (located in the master section) selects the "Solo In-Place" or "PFL" mode for the entire console.

Activating the solo switch in the pfl mode will send the prefader signal of the monitor section to the monitor/headphone outputs. In the solo in-place mode, all assigned signals in that channels post fader is heard, and all other channels are muted within the stereo mix. A solo indicator LED is fitted next to the solo switch. The SAFE switches in the channel prevent the SIP system from muting the channel.
The SAFE switch also prevent the mute computer from controlling the mute status of the channel. The MUTE lamp will have a higher intensity when both the SAFE switch and MUTE switch is activated to indicate that local muting is active.

The peak led in the channelmodule monitors signal level on 4 points in the channel.
The **MUTE** system is a special soft-muting system controlled either by the local mute switch or by the mute computer located in the master section. Low intensity muting will indicate that the mute function can be controlled by the mute computer. The SAFE switch prevents the mute being controlled by the mute computer.

3.6 THE FADER VCA SECTION

The Axion has a high quality 100mm ALPS fader, P&G faders are optionally available. The lineair fader controls a high quality dbx VCA which in turn can be controlled by a group master fader, if the channel is assigned to such a fader.

Assignment of the channel VCA fader is as follows.
By hitting the "VCA" select switch once, one of the led indicators 1-8 will light, indicating which VCA master is selected. By hitting the VCA select again the next in number VCA will be selected. If vca master 8 is on, the next selection will be vca nr 1 and so on. If you wish to de-select a channel from a vca master, simply push the select switch for more than 1 second and the assign switch will turn off indicating that the local fader has been selected for VCA control.

NOTE: the VCA max led indicates whether there is still gain left in the vca. If the VCA is controlled by a master VCA, it is very easy to "overload" the VCA by bringing up the master VCA fader above 0 dB, lets say to +5dB. If the channel VCA fader is already set to +5dB the maximum gain of 10dB is already given away, and no extra gain could be expected from the VCA, even when the channel fader indicates a further 5 dB of frontpanel gain.

CHANNEL METERS

The Axion is shipped with 25 segment ledbar meters in the standard configuation. The first LED in the bargraph is a power supply indicator.

3.7 INPUT MODULE IN AND OUTPUT CONNECTORS

Every channel has the following 3 pin XLR connectors at the back of the housing.

The balanced MIC input
The balanced LINE input

The ground compensated insert send
The balanced insert return
The balanced direct output

On the main printed circuit board there is a provision for mounting a mic input transformer allowing for ground separation.

Note: The default setting on the direct output is +4 dBu. A setting of -10 dBV can be chosen on the channel boards using jumpers. Any level between -80dB and +6 dBu is adjustable on the direct output.

A provision is made for inserting transformers on the line and group outputs.
There are also 3 pin locking headers for connecting multipin wiring to any of the above mentioned in/output connectors. The back-panel has room for optional multipin connectors

THE STEREO MODULE

4.0 THE STEREO MODULE DESCRIPTION

This stereo module is in most ways a copy of the channel input module with the exception of some extra items such as an image control and a pre/post solo system.

4.1 CHANNEL ASSIGN SECTION

The left and right signals from the balance control can be assigned in pairs to the groups and to the Left/Right buses. The assigment to the mono bus is a summed left/right signal post VCA.

4.2 BALANCE / STEREO WIDTH SECTION

The balance and stereo width controls are on concentrics for reasons of space limitations.
The balance control adjusts left/right imbalances, while the stereo width control changes the signal from mono fully left, over stereo (centre), to a huge stereo width at the full right position.
The balance control is on the top knob and the width control on the lower control. Both are centre detented.

4.3 GAIN SECTION

The input section consists of a stereo **GAIN** control. The gain control is a dual pot used to actively adjust the gain of two line amps (stereo input). The adjustment range is from - 20dB to + 20 dB.
A phase reverse switch reverses the left input connector pins to accomodate for improper phase relationships between the two inputs.

4.4 EQUALIZER SECTION

The three band stereo equalizer was designed for the type of equalization needed for stereo returns. The high frequency band is a shelving type at 12,000 Hz. Mid range is a bell type equalizer ranging from 200 Hz to 7000 Hz. The low frequency band is a shelving type at 60 Hz.

Each band has a boost or cut of 16dB. The whole equalizer is switchable in or out the circuit.

4.5 AUX SEND SECTION

The Aux send section has dual concentric pots to feed Aux busses 1&2 and 3&4 which is used for feeding stereo effects. Aux sends 1&2 and 3&4 are globally pre / post switchable in the master section. The mono switch related to Aux 1-4 has the following functions.
In the up position the left input signal will feed aux 1/3 and the right input signal will feed aux 2/4.
In the down position aux 1 to 4 will receive a summed left right signal to be sent to the aux 1-4 buses.
Aux send 5 to 12 will send a pre or post summed mono signal to the master aux 1 to 12 busses.

4.6 THE SOLO / MUTE SECTION

The peak led will indicate a +18dB level which is 4 dB prior to clipping on the following points.
Post, left /right line amps, and post, left / right equalizers.
The **SOLO** switches have two modes, PFL (pre fade listen) AFL (after fade listening)or a "destructive" stereo Solo-In-Place system. Master status switching (located in the master section) selects the "Solo In-Place" or "AFL/PFL" mode for the entire console.

Activating the solo switch in the afl or pfl mode will send the fader signal of the channel to the monitor/headphone outputs. In the solo in-place mode, the post channel panpot signal is heard, and all other channels are muted within the stereo mix buss. A solo indicator LED is fitted next to the solo switch. The SAFE switch prevents the SIP system from muting the channel.
The SAFE switch also prevent the mute computer from controlling the mute status of the channel. The MUTE lamp will have a higher intensity when the SAFE switch is activated to indicate that local muting is active.

The **MUTE** system is a special soft-muting system controlled either by the local mute switch or by the mute computer in the master section. Low intensity muting will indicate that the mute function can be controlled by the mute computer. The SAFE switch prevents the mute being controlled by the mute computer.

4.7 Fader VCA section

The Axion has a high quality 100mm ALPS fader, P&G faders are optionally available. The lineair fader controls the high quality dbx VCA's which in turn can be controlled by a group master fader, if the channel is assigned to such a fader.

Assignment of the channel VCA fader is as follows.
By hitting the VCA select switch once, one of the led indicators 1-8 will light, indicating which VCA master is selected. By hitting the VCA select again the next in number VCA will be selected. If VCA master 8 is on, the next selection will be vca nr 1 and so on. If you wish to de-select a channel from a vca master, simply push the select switch for more than 1 second and the assign switch will turn off indicating that the local fader has been selected for VCA control.
NOTE: the VCA max led indicates whether there is still gain left in the vca. If the VCA is controlled by a master VCA, it is very easy to "overload" the VCA by bringing up the master VCA fader above 0 dB, lets say to +5dB. If the channel VCA fader is already set to +5dB the maximum gain of 10dB is already given away, and no extra gain could be expected from the VCA, even when the channel fader indicates a further 5 dB of frontpanel gain.

4.8 Channel meters

The Axion is shipped with two 25 segment Ledbar meters in the standard configuration. The first LED's in the bargraph's are power supply indicators.

5.0 THE GROUP / MATRIX / VCA MASTER MODULE

This module consists out of three sections, the Matrix , the Group and the VCA section.
The upper part is the Matrix section. Above the Matrix section is the ledbar input select switch selecting either the Matrix outputlevel or the Group output level.

5.1 MATRIX INPUT SECTION

The matrix in the Axion can be fed from 12 sources and has 8 outputs.
The top control is the input for an external source or a second Axion console when two consoles are linked.
The Mono, Left and Right outputs can be mixed into the matrix outputs individually, pre or post main faders dependent upon jumper settings in the matrix module.

The eight group outputs can be mixed individually in the matrix.
These group outputs can be sent pre or post the group fader into the matrix.
The overall level of the matrix signal is controlled by a short travel lineair fader.
A “talkback enable” switch deblocks an already selected “talkback to matrix” routing in the communications module.
NOTE: The talkback signal is not changed by mute and / or fader settings in the Matrix output!
The PFL and MUTE switches speak for themselves.

5.2 GROUP OUTPUT SECTION

The group output mixes the assigned channel routing and controls this mix by a short travel lineair fader. The output of this fader can be assigned to the MONO buss, and via the pan-pot to the stereo busses. The group output can also be sent to the matrix. The signal can be taken pre or post the group fader.
The insert switch inserts a connected effects device into the signal path.
A “talkback enable” switch assigns an already selected talkback to group routing in the communications module.
NOTE: The talkback signal is not changed by mute and / or fader settings in the Matrix output!
The PFL and MUTE switches speak for them selves.

5.3 VCA MASTER SECTION

A unique feature of the Axion console is its ability to listen to DC subgroups. By hitting the AFL switch in the lower part of the Group/Matrix module you can listen to all the assigned channels to this VCA master. The signal is a stereo AFL signal from the channels panpots in a non destructive mode.
The mute switch is under control of the mute computer and totally mutes all assigned VCA faders to this master fader.
The Safe switch isolates the master VCA fader not to be controlled by the mute computer. The mute lamp will light more intens when used, when the safe switch is activated.
The eight VCA masters can be assigned to the grand master to have overall control over all VCA’s in the console.
NOTE: It should be noted that “overload” of the channel VCA’s is very easily created by bringing up both group master and grand master VCA faders above the 0 dB indication on the frontpanel.
The channel “max” VCA overload leds must be carefully examined from time to time.

6.0 INSTRUCTIONS FOR OPERATION

The AXION is designed to be the perfect answer for life sound reinforcement. In order to get more familiar with the Axion, we shall discuss the entire sound process.

LIFE

This is the beginning of a session. All input channels are placed in the mike mode by leaving the line switch in the up position if the microphone input is to be used in this channel. Phantom powering is applied if necessary. The EQ switch should be in the up position unless you require EQ on that mike. The signal flows through the fader and is available postfader to be routed by way of the assignment switches which can feed either the stereo output, the mono output, and or one or more of the group outputs.
The LED bargraph reads the incoming signal post line amp or post EQ, but prefader!.

Microphone / Line Gain

The amount of gain required may depend on the type of microphone being used, the sound pressure level, and the distance between the sound source and microphone. When the line switch is activated, the same gain control varies the gain of the separate electronics for the balanced line input. The "phase" switch affects both the mike and line inputs.

After plugging in a mike or line signal, turn the gain control clockwise until a "0″ output level is reached on the related channel meter. Now slide up the channel fader to "0″. If the signal <u>source</u> gets louder or softer, it may be necessary to re-check this setting. The volume will also fluctuate if you boost or cut the equaliser section.

Multiple Modules Assigned to One or More outputs

When more than one microphone or line signal has to be processed there are basically two ways of doing this. You choose an audio subgroup or you use the VCA subgrouping facility.
Simply route to one of the 8 subgroups by activating a channel routing switch on as many input modules as required. Decide on which group you wish to sum these signals and activate the related numbers
. The group metering will show the subgroup level which can be changed overall by the short travel group fader. In order to monitor these groups on the group modules, simply push the pfl switch in the group modules.
An alternative way of routing is by way of the VCA grouping system. Assign the channels which you want to control as a group to one of the eight VCA masters, by momentarely hitting the vca

switch which will step through all vca master assign leds. If you have assigned your channel to the group VCA, the group master VCA will control the channel VCA from now on.
The audio will still be processed through the audio summing amps. This could be the mono, stereo and or audio groups.

Insert Channel / Group

For high dynamic range types of inputs, a signal processor such as a compressor / limiter can be inserted in the channel or in the group insert if an entire group signal needs to be processed.

Stage monitoring

During life performaces it is essential that the talent hear an independent mix of what the engineer and audience are hearing. Monitor mixes are usually derived from pre-fader auxilliaries. In the Axion aux 1 & 2 and 3 & 4 are ideal for this purpose. Put the Aux 1-4 globally to pre fader in the master section. And set up the required mix as requested by the talents. Mostly a monitor console will go with the FOH Axion console to create the stagemix. We will not go into detail concerning stage mixes.

Effect Sends

All unused aux sends can be used to send signals to signal processors such as the D&R "Qverb" 16 bit digital reverb, effects processors, and digital delays. The aux sends are usually post-fader in order that the right balance between untreated and treated signals is maintained however, it is possible to switch to pre-fader.

Effects Returns

In modern life sound , there is a demand for many effect returns and inputs for MIDI related gear. For that reason D&R has designed the Axion with full functioning stereo effects return modules. See section 4.0 of this manual for a complete description of this module.

In life sound, communication is essential for a succesfull performance of all artists. The internal communication system makes it possible to do "last minute" sound checks.
You have set up all mikes and just before the show starts, there has to be some changes in mike settings.
You have your main faders down and you are sending CD music to the audience through the 2 track fader with or without EQ. By hitting the PRE FDR switch in the monitor section you are able to listen to all the mikes assigned to the main outputs and or to the MONO output. (NOTE: all these input signals can be summed).

Either the stage engineer or you can call eacg other by way of the CALL A or B switch, to contact each other. You as main engineer assigns both the COMM "A" switch and the "COMM to PHONES" switch to be able to hear what 's going on on stage.

You communicate by hitting the "COMM" switch momentary in its "down" position or permanently in its "up" position.

The stage engineer tells you for instance that he has to change mikes and you tell him which channel he has to plug in the mic. Now you hit "PHONES to COMM". in the communications module and solo the mentioned channel to listen what's going on in that specific channel.
Both you and the stage engineer can hear that channel in your headphones , while the audience is still listening to CD music.

By switching off the "COMM to PHONES" you as mix engineer can continue in checking other items while the stage enginer still got his solo'ed channels in his headset. This is a situation you can easily handle with the AXION. There are many more practical features to be experienced during life performances and sound checks in your AXION console.

7.0 INSTALLATION - ELECTRICAL

Local Electrical Voltage

Before connecting the Axion, check the AC supply voltage setting by looking at the sticker on the back of the rack mount power supply. This sould be 110V for use in areas with an AC supply between 100V and 120V, and 220V for use in areas with an AC supply between 220V and 240V.

The main fuse should be 10 amp, 20mm (fast blow) for 110V service, and 6.3 amp, 20mm (fast blow) for 220V service. If one or more of the power supply LED indicators should go out, turn off the power supply and check the fuses on the back panel of the rack-mount power supply. After replacing a blown fuse with the correct size and rating, turn the power supplies on and check the three LED indicators. If you are still missing one or more of the power rails, turn off the power supply and call the D&R Technical Support Department. DO NOT REPLACE THE FUSE WITH ANY OTHER TYPE AS THIS CAN BECOME A SAFETY HAZARD AND WILL VOID THE WARANTY.

Electrical Wiring

To take full advantage of the excellent signal to noise ratio of the Axion, it is necessary to read this part of the manual carefully.

Hum, radio frequency interferance, buzzes and instability are often caused by improper wiring and poor grounding. Sometimes the incoming electrical ground is inadequate and a dedicated ground would need to be installed for the audio equipment.

Your local electric power company will provide you with all local electrical codes and safety regulations. There are some grounding rules to follow. All signals in a life sound situation are referenced to ground. This ground must be clean and free of noise. A central point should be selected as the main grounding point and all grounds should originate from this point. This is commonly referred to as a "star ground system".

In some instances electrical contractors will daisy chain ground connections. This is unsuitable for a life performance. Ideally, run a separate ground wire from each outlet and a separate ground wire for each piece of equipment. A separate wire from each equipment rack to the dedicated ground point is useful in cases where AC outlet grounds are not satisfactory.
The dedicated ground point should be located at the rear of the console or equipment rack. Separate and identify "clean" and "dirty" AC outlets. Use clean outlets for audio equipment and the dirty ones for lighting, air conditioning, freezers etc. Do not intermix these two types of outlets. AC interferance can be greatly reduced by using an isolation transformer (Juice Goose) to power clean outlets. Ground this transformer directly to the dedicated ground point or as close as possible to the incoming ground.

All equipment should be physically located as far as possible from the main breaker panel. Unbalanced equipment may require isolation from the equipment rack so that ground loops are avoided.

8.0 INSTALLATION - AUDIO

8.1 Interface Monitor Levels

The Axion in its standard configuration can interface with all available equipment.

Attention concerning Monitor output must be noted. This output delivers a nominal +4 dBu level which is sometimes too high for power amps rated at 300mV sensitivity for full output. In some instances an input attenuator at the power amp's input is required to reduce this +4 dBu level by up to 12 dB. Contact the D&R Technical Support Department for details.

8.2 The Initial Hook-Up

First connect the rack-mounted power supply to the console. All faders, and effect returns must be in the "down" or "off" position. In order to ensure the best signal to noise ratio for your system, the next steps should be performed in the order they are printed.

Connect the Monitor outputs (located on the master module backplate) to the inputs of your control room speaker power amps. Now turn on the console power supply and then turn the power amp on and check for any hum, buzz, or interferance. Slowly turn the Monitor control clockwise until it is wide open while listening for excessive noise. You should only hear a faint "hiss". If everything is O.K., continue. If any hum or excess noise is present, stop and try different ground and shielding arrangements until the system is clean.

NOTE: MAKE SURE THAT YOU CHECK FOR HUM OR NOISE AS EACH INPUT OR OUTPUT IS CONNECTED.

8.3 SHIELDING AND GROUNDING OF AUDIO EQUIPMENT

The shield of any audio cable connection should be connected at one end only. If not, ground loops and high frequency cross-talk could result. Connect the shield as a general rule to the signal source (output) of anything. In high RF areas it is wise to ground the other end of the shield through a 0.01 microFarad capacitor. This will ground the RF but will not affect audio frequencies.

8.4 TYPICAL INTERFACE SITUATION TABLE

Output	Input	Connect shield at:
Unbalanced	Unbalanced	Output
Unbalanced	Balanced	Output
Unbalanced	Differential	Output
Balanced	Unbalanced	Input
Balanced	Balanced	Output
Balanced	Differential	Input
Differential	Unbalanced	Output
Differential	Balanced	Output
Differential	Differential	Output

Use the above table to interface your Axion to any external equipment such as multi-track machines, signal processing, and power amps. Balanced (in the above illustration) means transformer balanced while differential means electronically balanced. There are some cases which net better results in practice. Connect one circuit at a time and check for hum or noise. When connecting balanced microphones, use two conductor shielded audio cable and connect both conductors and the shield at both ends.

When connecting line level cables, use two conductor shielded cable and follow the instructions in the interface table. The only exception to these rules is with patch cords. These grounds are tied together in the console. We realize that the correct interfacing of all different equipment is difficult, but once properly installed the system will be clean and noise free.

It is important to understand the term balanced. Balanced does not mean the input or output is professional, the single factor that normally determines whether something is professional is the level of the input or the output. +4 dBu is considered professional. -10 dBv is considered semi-professional. Because many semi-professional tape machines are built to professional specifications, D&R builds into the Axion console the ability to interface with both levels.

8.5. MASTER SECTION MODULE CONNECTORS

All connectors on the Axion console are of the XLR 3 pin type.

All audio connectors are wired as follows

Pin 1 = ground
Pin2 = Hot (in phase)
Pin3 = Cold (out of phase)

The lamp connectors are differently wired and will accept a 12 volt 5 watt lamp as a maximum.
Pin 1 =ground
Pin2 = + 12 volt
Pin3 = not connected

We recommand the use of "Littlite" type 18XRA, 18″

NOTE: The "Littlite" lamp wiring has to be modified to match the wiring of the consoles lighting female XLR's. The reason is that if we did not modify this you could easily destroy a microphone connected to these XLR sockets.

8.6 CONNECTING / CONFIGURING OF THE CHANNEL MODULE

The channel backplate has the following 3 pin XLR connectors:

Mic input
Line input
Insert send
Insert return
Direct output.

All connectors are of the XLR 3 pin type and have the international standard wiring whereby
pin1 =ground,
pin2=hot,
pin3=cold.

On the connector pcb's there are Molex 3pin locking connectors to be used to wire multipin connectors to some or all of the in/outputs. There are also 4 pin headers to interface with optional transformers for the Line and Direct output. The optional mic transformer should be placed on the main pinted circuit board.

8.7 JUMPER SETTINGS ON THE CHANNEL

The following changes in signal flow could be made by jumper settings on the printed circuit channel boards.

FUNCTION CONN. TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Direct output level (conn-2)	+4 dBu	-10dBV
Direct output (conn-3)	post channelfader	Pre channel fader
Meter select (conn-1)	line output	post EQ output
Aux pre select (conn-4)	post Mute/post EQ	pre Mute, pre EQ
PFL (conn-10)	PFL-ON	PFL-OFF
AFL (conn11)	AFL OFF(no jumpers)	AFL ON (both jumpers)

The output level of the balanced output of the channel can be adjusted by the VR1 trimmer, which ranges from infinity up to +6 dBu if necessary.
Channel and group inserts (sends and returns) are used to patch (pre-fade, pre EQ, post High pass filter in the channel) into the channel or group, any signal processing equipment such as compressors, limiters, equalisers etc.

The LINE INPUT is used for plugging in the outputs of digital reverbs, digital delays, drum machines, samplers, keyboards, CD players, cassette machines and any machines with line level outputs.
The **MIC INPUT** is used for plugging in all types of microphones or direct boxes. This is an active balanced input using the latest circuit technology available today. Each channel module has 48 volt phantom powering indivually switchable.

DO NOT USE EXTERNAL PHANTOM POWER AND THE POWERING IN THE CONSOLE AT THE SAME TIME!

8.8 JUMPER SETTINGS ON THE STEREO CHANNEL

The following changes in signal flow could be made by jumper settings on the printed circuit stereo channel boards.

FUNCTION CONN TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Aux 3 (conn-7)	Left/summed mono	follows left only
Aux 4 (conn-8)	right/summed mono	follows right only

Channel inserts (sends and returns) are used to patch (pre-fade, pre EQ, post High pass filter in the stereo channel) into the stereo channel any signal processing equipment such as compressors, limiters, equalisers etc.

8.9 JUMPER SETTINGS ON THE GROUP/ MATRIX MODULE

The following jumper settings are available on the group matrix modules.

FUNCTION CONN TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Mono to matrix (conn-8)	Pre fader to matrix	Post fader to matrix
Right to matrix (conn-7)	Pre fader to matrix	Post fader to matrix
Left to matrix (conn-9)	Pre fadeer to matrix	Post fader to matrix
Group to matrix pre/post group mute (conn-1)	Post group mute	Pre group mute

The following levels can be adjusted.

Matrix output levelfrom infinity to +6dBu (+4dBu is the default setting)
Group output level from infinity to +6dBu (+4 dBu is the default setting)

8.10 JUMPER SETTINGS AND LEVELS ON THE MASTER MODULES

MODULE AUX 1/7

The following levels can be adjusted from infinity to +6dBu (+4dBu is the default setting)

Aux 1, Aux 7,

MODULE AUX 2/8

FUNCTION CONN TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Phantom to Listen mic	No phantom power	+48Volt phantom power

The following levels can be adjusted from infinity to +6dBu (+4dBu is the default setting)

Aux 2, Aux8, Main Left, Main Right, PFL output, AFL left, AFL right, Monitor left, Monitor right.

MODULE AUX 3/9

FUNCTION CONN TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Phantom to T.B. mic	No phantom power	+48Volt phantom power

The following levels can be adjusted from infinity to +6dBu (+4dBu is the default setting)

Aux 3, Aux9, Main Left, Main Mono, Talk Back direct output,

On the frontpanel the side tone levels of the communication system can be adjusted.

MODULE AUX 4/10

FUNCTION CONN TYPE	DEFAULT SETTING	ALTERNATIVE SETTING
Rec out left, (conn-4)	+4dBu	-10dBV
Rec out right, (conn-5)	+4dBu	-10dBV
2Track A left in, (conn-6)	+4dBu	-10dBV
2 TrackA right in, (conn-7)	+4dBu	-10dBV
2 Track B left in, (conn-8)	+4dBu	-10dBV
2 Track B right in, (conn-9)	+4dBu	-10dBV

The following levels can be adjusted from infinity to +6dBu (+4dBu is the default setting)

Aux 4, Aux 10, Record out left, Record out right,

MODULE AUX 5/11

NO JUMPERS ON THIS MODULE

The following levels can be adjusted from infinity to +6dBu (+4dBu is the default setting)

Aux 5, Aux 11,

Adjustments can be made on the following trimmers

VR5	distortion oscillator
VR6	distortion oscillator
VR7	output level oscillator only
VR4	output level pink noise generator only

MODULE AUX 6/12

NO JUMPERS ON THIS MODULE

The following levels can be adjusted from infinity to +6dBu
(+4dBu is the default setting)

Aux 6, Aux 12,

9.0 LINKING TWO AXION CONSOLES

It is possible to link two or more Axion consoles by "daisy chaining the male and female "Link" connectors by "ordinairy" balanced mic cables.
NOTE: It is advisable to cut the shield at the master console side!
As soon as two or more consoles are linked the solo system is active on all consoles. Any solo switch in any of two or more consoles, always activate the monitoring in all consoles. So is the case with the destructive solo in place system. Any depressed solo switch will always put all other channels into mute (if the consoles are linked).
The usual way of linking two (or more) consoles is to connect all Groups/Matrixes/Auxes/Left/Right/Mono/PFL/AFL-L/AFL-R from one console to the next to create a master/slave configuration whereby all signals from the "slave" enter the "master" console.

NOTE:

It is very important to ground/connect the two consoles only at their starground point and have the master slave conectors only have their shields connected at the output of the slave console.
Power supplies can be grounded both if necessary. If ground loops occur, remove the slave power supply grounding.

Every installation is different, so it is important to be as consequent as possible with reason, you must follow logic and only one system of grounding for the best results.

Troubleshooting and servicing

10.0 Troubleshooting

It is essential to study the signal flow chart carefully, only then can you hope to isolate problems. By tracing the signal from input to output XLR's, it is possible to locate a problem. If for any reason you are unable to isolate a problem, contact the D&R Technical

Support Department for advice. If the problem cannot be corrected over the phone, D&R will despatch a replacement module (ground freight prepaid) the same day. Most problems can be found using logical thinking and simply replacing socketed integrated circuits.

10.1 Removing a Module

The Axion is a complex piece of equipment and some understanding of its internal layout is necessary before removing a module.
An input module has wiring to the master section and backplates and the starground wire. Every 8 modules there is a power supply wiring. All of these wires must be removed before withdrawing a module from the console. Each module has computer grade connectors for ease of the disconnect.

Turn off the power supply. Remove the upper strip which covers the module and the plexiglass.
Remove the two module retaining screws. It is often easier to also remove the modules positioned left and right of the module under test. It is now possible to carefully lift the module until the module wiring can be unplugged. Now remove all flat cables from the bottom of the module PCB. At this point extender cables (if ordered) can be connected.
The master sections can be removed from the frame in the same way.
Because of the many flat cables on the bottom of the master section, it is wise to remove all retaining screws from all master sections, and remove the blank module on the left/right side of the master section. This will allow all the master modules to be moved slightly without unplugging all the flat cables. A qualified service technician will be able to service the modules in this way.

Dear Axion owner,

In this manual we have tried to give you an overview of all that the Axion has to offer. If you have any questions, do not hesitate to contact us or the D&R USA customer support department. With the Axion series there is no limit to your creativity. We wish you many years of enjoyable mixing.

Best regards,

Duco de Rijk
PRESIDENT D&R, HOLLAND

This manual was written by Duco de Rijk (D&R Holland) and Paul Westbrook (D&R USA). We hope you will find it to be useful and easy to understand. As always, we are open to any suggestions about this manual or any D&R product.

PRODUCT SAFETY

This product is manufactured with the highest standards and is double checked in our quality control department for reliability in the "HIGH VOLTAGE" section.

CAUTION

Never remove any panels, or open this equipment. No user servicable parts inside.
Equipment power supply must be grounded at all times.
Only use this product as described, in user manual or brochure.
Do not operate this equipment in high humidity or expose it to water or other liquids.
Check the AC power supply cable to assure secure contact.
Have your equipment checked yearly by a qualified dealer service center.
Hazardous electrical shock can be avoided by carefully following the above rules.

EXTRA CAUTION FOR LIVE SOUND

Ground all equipment using the ground pin in the AC power supply cable. Never remove this pin. Ground loops should be eliminated only by use of isolation transformers for all inputs and outputs. Replace any blown fuse with the same type and rating only after equipment has been disconnected from AC power. If problem persists, return equipment to **qualified service technician**

PLEASE READ THE FOLLOWING INFORMATION

Especially in sound equipment on stage the following information is essential to know.
An electrical shock is caused by voltage and current, actually it is the current that causes the shock.
In practise the higher the voltage the higher the current will be and the higher the shock.
But there is another thing to consider and it is resistance. When the resistance in Ohms is high between two poles, the current will be low and vica versa.
All three of these; voltage, current. and resistance are important in determining the effect of an electrical shock.
However, the severity of a shock primarily determined by the amount of current flowing through a person.

A person can feel a shock because the muscles in a body respond to electrical current and because the heart is a muscle it can affect, when the current is high enough. Current can also be fatal when it causes the chest muscles to contract and stop breathing. At what potential is current dangereous.
Well the first feeling of current is a tingle at 0.001 Amp of current. The current between 0.1 Amp and 0.2 Amp is fatal.
Imagine that your home fuses of 20 Amp can handle 200 times more current than is necessary to kill. How does resistance affect the shock a person feels. A typical resistance between one hand to the other in "dry" condition could well over 100,000 Ohm.
If you are playing on stage your body is perspiring extensively and your body resistance is lowered by more than 50%.
This is a situation in which current can easily flow. Current will flow when there is a difference in ground potential between equipment on stage and in the P.A. system. Please do check if there is any potential between the housing of the mikes and the guitarsynth amps, which will be linked by your body on stage. Imagine, a guitar in your hand and your lips close to the mike! A ground potential difference of above 10 volts is not unusual, in improperly wired buildings it can possibly be as high as 240 volts.
Allthough removing the ground wire sometimes cures a system hum, it will create a very hazardeous situation for the performing musician.
Always earth all your equipment by the grounding pin in your mains plug.
Hum loops should be only cured by propr wiring and isolation input/output transformers.

Replace fuses always with the same type and rating after the equipment has been turned off and unplugged.
If the fuse blows again you have an equipment failure, do not use it again and return it to your dealer for repair.

And last but not least be carefull not to touch a person being shocked as you, yourself could also be shocked.
Once removed from the shock, have someone send for medical help inmediately
Aways keep the above mentioned information in mind when using electrically powered equipment.

Conformity statement according to ISO/IEC Nr. 22 and EN 45014

Name Manufacturer	**D&R Electronica Weesp b.v.**
Addres manufacturer	**Rijnkade 15B, 1382 GS Weesp, The Netherlands**

declares that this product

Name product	**AXION series**
Modelnumber	**n.a.**
Produktoptions	**All**

passed the following product specifications:

Security	**EN 60950: 1988 +A1, A2**
EMC:	**CISPR-22: 1985 / EN 55022: 1988 class B (*) EN 50082-1: 1992 IEC 801-2:1991 / prEN 55024-2:1992 - 3kV CD, 8kV AD IEC 801-3:1984 / prEN 55024-3:1991 - 3 V/m IEC 801-4:1988 / prEN 55024-4:1992 - 0.5kV signalcables, 1 kV powercables.**

Extra information:

The product passed the specifications of the following regulations;

Low voltage 73 / 23 / EEG
EMC-regulations 89 / 336 / EEG.

(*) The product is tested in a normal users environment.

D&R