ORION

USER MANUAL

Introduction and Product Overview

1.0 The Chassis system - description

1.0 The Orion chassis system

2.0 Master section - description

2.1 Solo section
2.2 CRM section
2.3 Oscillator section
2.4 Communications section
2.5 Auxiliary send masters
2.6 Studio section / Phones section
2.7 Master inputs / outputs
2.8 Master meters / Channel meters / Phase meter

3.0 "LCRS" in-line module - description

3.1 Input section
3.2 Equalizer section
3.3 Auxiliary send section
3.4 Monitor section
3.5 Floating Subgroup section
3.6 Channel assign / Fader section
3.7 Automation
For inputs / outputs - see 4.7

4.0 Standard in-line module - description

4.1 Input section
4.2 Equalizer section
4.3 Auxiliary send section
4.4 Floating Subgroup System
4.5 Monitor section
4.6 Channel assign / Fader section
4.7 Inputs / outputs

5.0 Dual Stereo Aux Return module - description

5.1 Input section
5.2 Equalizer section
5.3 Aux send section
5.4 Floating Subgroup System
5.5 Channel assign / Fader section
5.6 Inputs / outputs

Letter from the Prez

Dear Orion owner,

Thank you for selecting the D&R Orion series.

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

We value your suggestions and would appreciate you taking the time to complete and return the questionnaire included at the front of this manual (once you become familiar with your Orion). We listen and learn from your comments and you can be assured that our research and development department will take your comments very serious.

Built to the highest standards, we are confident that the Orion will help produce "state of the art" recordings for many years and wish you much success.

With kind regards,

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

RAISING
THE
STANDARD

Orion Recording Console

The D&R Orion series is a balanced, 8 or 16 buss, in-line format recording and mixing console designed to take the central role in a recording, mixing, or post production facility.

The Orion is completely modular and can be configured to precisely suit your particular system requirements. Due to the fact that all inputs and outputs can be connected using the individual module and master section connectors, the Orion patchbay is entirely optional. When the Orion patchbay is installed, the Orion can be interfaced using 25 pin sub D multipin connectors. If you have your own patchbay or perfer to wire your own, individual input / output connectors (rear of meterbridge) can be used for the interface. If you ordered your Orion short-loaded and without the patchbay, you may install the patchbay at a later date.

To become completely familiar with your Orion 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 Orion including; installation, operation, and servicing.

If we can be of any assistance to you, don't hesitate to call.

Head Office / Factory:

D&R Electronica B.V.
Rijnkade 15B
1382 GS Weesp
The Netherlands

Tel: (-) 31 2940 18014
Fax: (-) 31 2940 16987

1.0 The Orion Chassis System

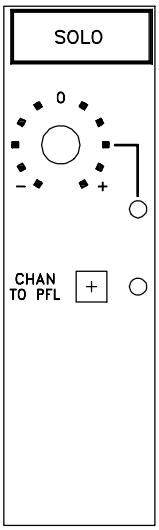
The Orion is available in three frame sizes; 30, 38, and 51 (larger frames can be special ordered). The basic frame has four blank modules, two are located on the extreme left and right of the frame and two more are on the left and right of the master section. The extreme left and right blanks cannot be used for input modules as they conceal mechanical construc- tions, however, the blanks on either side of the master can be replaced with input modules. A heavy duty rack-mount power supply, master modules, and internal cable harness are included.

***Note**: Due to the LCRS module and standard in-line module being the in-line design format, we will refer to both as in-line modules. As always, with D&R you can pick and choose how many of any of the three modules you want and where you would like them in the frame.*

The frame 30 will fit 32 input modules (in-line, or dual stereo modules), 4 master modules, and 2 blank modules. The standard configuration has (from left to right) 24 in-line modules, 4 master modules, and 6 duel stereo return modules. Custom configurations are available at no extra charge.

The frame 38 will fit 40 input modules (in-line or dual stereo modules), 4 master modules, and 2 blank modules. The standard configuration has (left to right) 24 in-line modules, 4 master modules, 8 in-line modules, and 6 dual stereo return modules. Custom configurations are available (no charge).

The 51 position frame will fit 40 input modules (in-line or dual stereo modules), 4 master modules, patchbay, and 2 blank modules. The standard configuration has (from left to right) 24 in-line modules, 4 master modules, 8 in-line modules, 6 dual stereo return modules, and patchbay. Custom configurations are available at no extra charge. The 51 position frame can accept up to 54 input modules if the patchbay is not utilized.
Both the master section and patchbay can be installed wherever most suitable, but the request must be made at the time of ordering. If your requirements are for your patchbay to be remotely mounted, ask D&R for a custom quote. The Orion is available with or without the pedestal base and stabilizing beam. If ordered with console, the pedestal is not assembled for shipping. Assembly takes approximately twenty minutes.

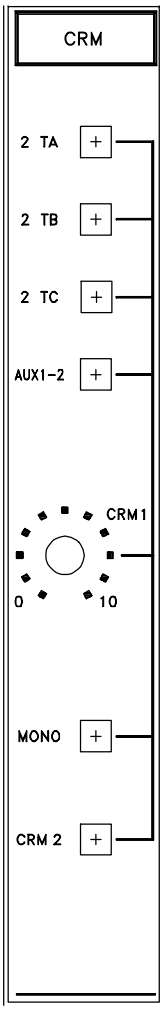


2.0 Master section - description

The Orion has four master modules mounted in a single front panel . 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 volume control with a center detent for nominal levels and a **CHAN**nel to **PFL** (pre-fade listen) switch with an LED indicator mounted. When this switch is in the up position, all solo switches on the input modules are in the "stereo-in-place" mode when activated. With this switch in the down position, any solo depressed would be in pre-fade listen mode. An LED indicator is also fitted next to the solo level control to show when a solo circuit is activated.



2.2 CRM section

The **CRM** (control room monitor) section contains the electronics for monitoring all signal paths in the Orion.

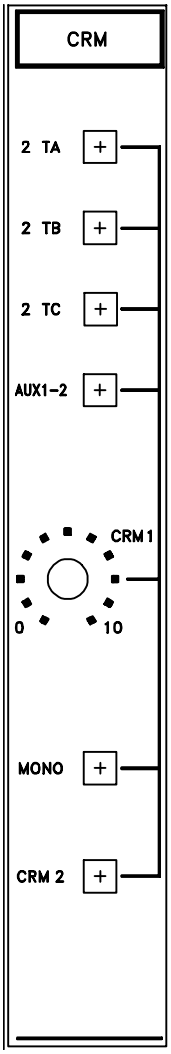
CRM Source switching

From the top of this section, there are the four CRM signal sources. With all these switches in the up position, the CRM will monitor the stereo main outputs which is the sum of all the monitor sections (assigned on LCRS), Left / Right routing switches in the input / output modules, and all (assigned) stereo inputs from the dual stereo modules. The Orion has three dedicated balanced +4 dBv or -10 dBu tape machine returns which can be wired to the outputs of stereo master machines, cassette machines, CD players, or DAT recorders or to the Orion patchbay. Utilizing any of these switches enables playback or post tape monitoring of a master mix.

Aux 1-2 (7-8 if fitted with LCRS modules) can be monitored in stereo to allow the building-up of a stereo cue mix for the headphone system through the control room monitors.

CRM level

The **CRM** level controls the output level to the control room monitors amps. The Orion has two CRM systems intended for use with large monitors and nearfield monitors which are switchable via the **CRM 2** switch. We advise that nearfield monitors be wired to the CRM 2 output since the communica- tion takes place over this monitor while dimming the main monitor.



MONO

The Mono switch allows the user to listen for any out-of-phase signals or merely used for monitoring your mix in mono.

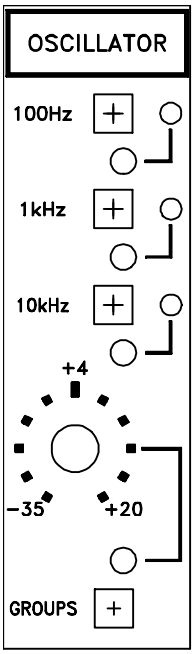
Faders

Located in the bottom of the left two master modules (not drawn) are two 100mm mono faders which control the main stereo mix busses.

2.3 Oscillator section

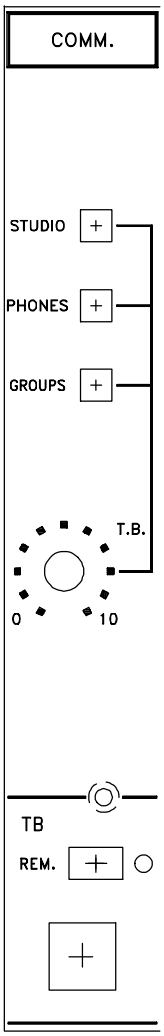
A three frequency, low distortion, phase shift type oscillator is fitted. The three frequencies are; 100Hz, 1kHz, and 10kHz. Each frequency has its own front panel alignment trimmer and overall level control to adjust the oscillator for precise alignment of the console and tape machines. The level ranges from -35 dB to +20 dB with a detented mid-position calibrated at +4 dBv. The oscillator is routed to the stereo mix busses and switchable to the GROUPS.

NOTE: The CRM will dim 20dB when the oscillator is active.



The meters on the Orion are peak reading meters and will read -6 dB when a sine wave with a +4 dB output level is sent to the meter. Measuring the +4 dB output level of the channel or master with a AC voltmeter would produce a 1.22 volt reading. The LED meter would actually be reading -6 dB on the scale. When monitoring "program" material, you will see higher levels on the peak reading meters. A pair of analog VU meters are available as an option to fit alongside the master peak meters.

Note: When aligning your tape machines to the peak meters on the console, do not set the peak meters to "0". The meters should be set to -6 on the console meters. If analog VU meters are used on the master section, they would be set to "0".



2.4 Communications section

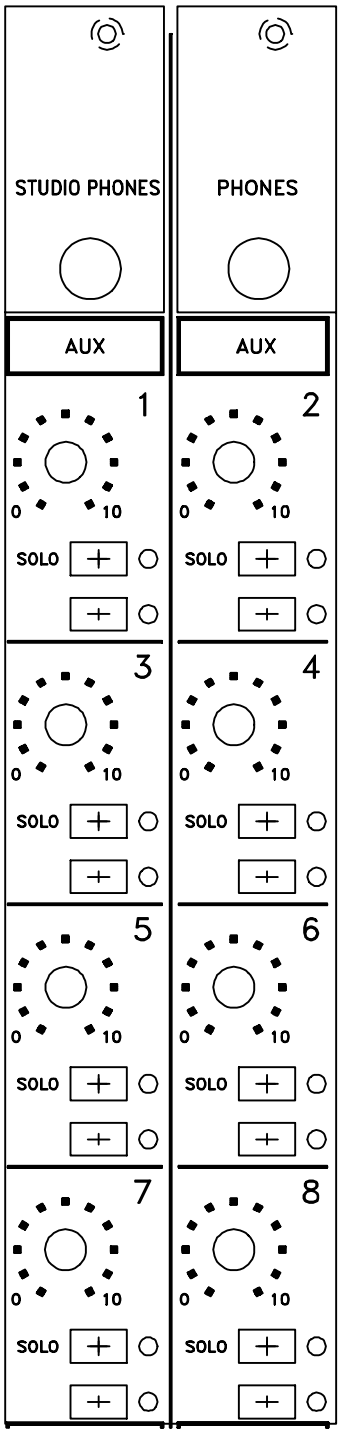
The communications section allows all communication functions from the control room to the STUDIO, PHONES, and GROUPS. Each buss has its own assignment switch and is feed via the T.B. control and built-in condenser mic.

The T.B. REM. (remote) switch is for locking in the talkback allowing the engineer or producer to carry on a two way conversation with the studio from remote areas in the control room. When depressing this push-on push-off switch, you can communicate without holding down the talkback switch.

The talkback switch is a momentary type which allows you to talk to the head**PHONES**, **STUDIO** speakers, and 8 GROUP outputs as well as the "Floating Subgroup System". The stereo phone jack sockets located above the aux master sections are for checking (on stereo headphones) the mix being sent to either output section.

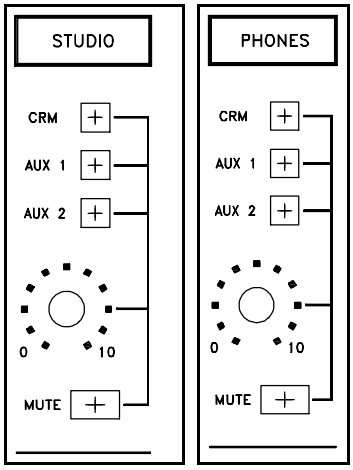
2.5 Auxiliary send masters

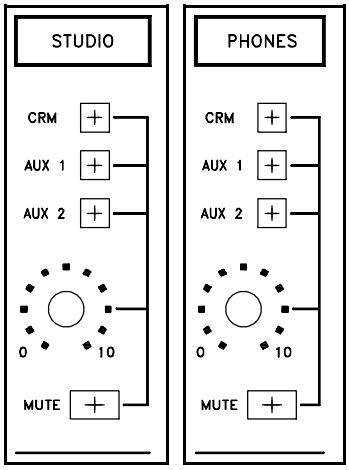
The eight auxiliary master sections are identical in function. Mounted in the first module are aux sections 1, 3, 5, & 7 and mounted in the second module are aux sections 2, 4, 6, & 8. Each Aux master pot controls the sum of all channel or monitor aux send busses. The solo switch sends the (post fader) aux buss signal to the control room monitors. The associated LED lights indicating the activated solo switch. When using any aux send buss, turn the aux master full clockwise for the correct amount of output. If the input level of your effects device seems to be low, make sure the input gain on any module you are sending from is adjusted correctly. In order to adjust this control, refer to section 4.1 of this mauual.



2.6 STUDIO and PHONES section

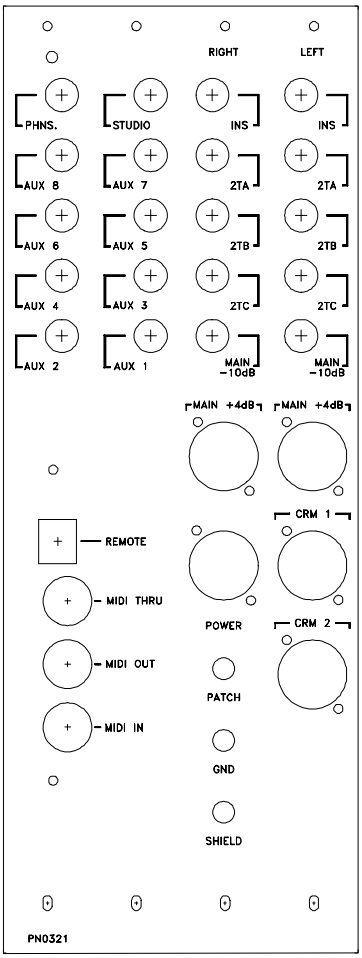
The STUDIO section and PHONES section performs in a similar way to the CRM module, however, the signals are sent to the studio or phones amp from the CRM or Aux 1 & 2. If you press only the Aux 1 switch, you will hear only the Aux 1 buss (in mono) through the studio speakers (or the headphones) and if you press only the Aux 2 switch, you will hear only the Aux 2 buss (in mono) through the studio speakers (or headphones), however, if you press both Aux 1 and Aux 2 switches you will hear Aux 1 in the left studio speaker (or headphone) and Aux 2 in the right studio speaker (or headphone). When pressing both switches, you can have a stereo mix playing through the studio speakers (or stereo headphones).





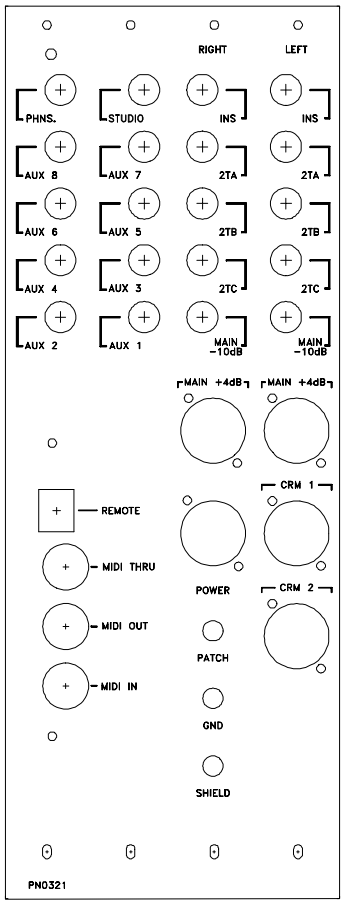
Also fitted in this section is a mute switch and level control with level ranges up to a balanced +4 dB nominal level (+26 dB maximum). The mute switch is used to mute the output of the studio or phones module. The studio output section can be used for feeding a completely separate mix to a second set of studio headphones.

2.7 Master Inputs/Outputs



The Orion has two methods of interfacing with external equipment such as two track master machines, signal processors, headphone amps, and power amps. Interfacing is possible using the connectors on the master back panels, through 25 pole sub D male connectors (if utilizing the optional patchbay), or a mixture of both. Listed below are all inputs and outputs for the four master modules. The next four drawings and paragraphs describe and explain the backplate and all connectors.

Master module # 1 & 2 (CRM & SOLO and OSCILLATOR & COMM.) This double backpanel houses (from top to bottom) the right and left master insert sends & returns, two track A, B, & C right and left playback inputs (from two track machines), -10 dBv right and left outputs, and +4 dB balanced right and left outputs fitted on male XLRs. Also fitted on male XLRs are CRM 1 & 2 outputs. The ground terminals and power supply connector are located on this backplate as well.



Right and left master insert in and out.

Right and left two track A (playback)

Right and left two track B (playback)

Right and left two track C (playback)

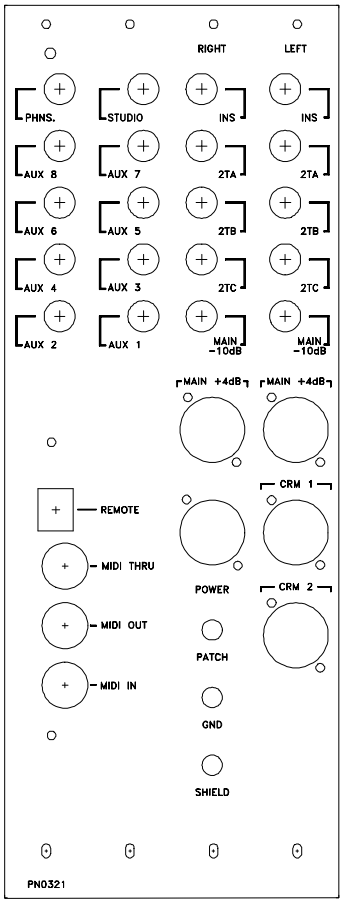
Right and left -10dBv main outputs

Right and left +4 dB balanced main outputs

Power supply connector / Control room 1 output

Ground terminals and Control room 2 output

Master module # 3 & 4 (Aux masters 1 through 8 and Studio & phones sections). The connectors on this module backplate are the studio & phones stereo outputs and the balanced Aux 1 through 8 master outputs. The Phones jacksocket is a stereo 1/4" output which should be wired to the inputs of a stereo power amplifier intended to drive the headphones in the studio. The Studio jacksocket is a stereo 1/4" output which should be wired to the inputs of a stereo power amplifier intended to drive the Studio playback speakers.



Phones output and Studio output

Aux 8 output and Aux 7 output

Aux 6 output and Aux 5 output

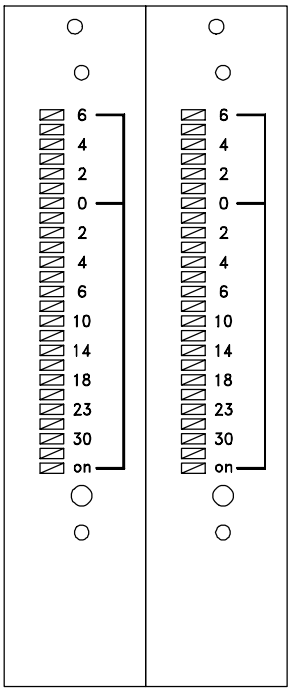
Aux 4 output and Aux 3 output

Aux 2 output and Aux 1 output

Automation remote control

MIDI in, out, and thru din sockets

Aux sends 1 through 8 are used for sending signals within the console to eight different signal processors such as digital reverbs. These outputs are balanced (ground compensated).



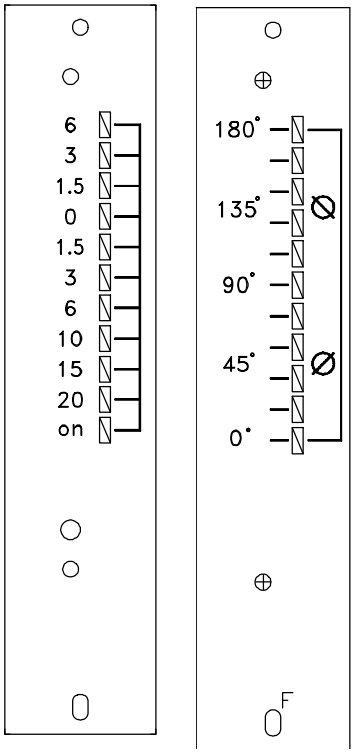
2.8 The Master, Channel, & Phase Meters

Master Metering

The Orion master is fitted with peak reading, high resolution, LEDbar 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.

NOTE: Peak reading meters give a reading 6 dB below the actual level when using a sine wave. For example, +4 dBu at the output connectors would give a reading of -6 dB on the meter using the oscillator.

If U.K. reading ledbars are ordered, they will have attack and release times of 300msec. and a +4 dBu level on the output connectors will give a 0 dB reading on the meter.



Channel metering

Each in-line channel is fitted with peak reading, eleven segment LEDbar meters mounted in the meter bridge. The attack and release times conform to world standards. The attack is 10msec. for a 20 dB range and the release is 1.5msec.

NOTE: Peak reading meters give a reading 6 dB below the actual level when using a sine wave. For example, +4 dBu at the output connectors would give a reading of -6 dB on the meter using the oscillator.

Phase Meter

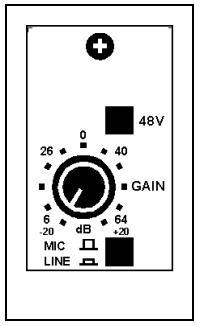
Fitted in the meterbridge is a phase corelation meter. This meter is used for checking for phase cancellations in the stereo mix or when miking anything in stereo. It is normal for your meter to display up to 90 degrees when monitoring in stereo.

3.0 The LCRS Module - description

The Orion LCRS module is a basic input / output design (in-line) whereby all signal flow takes place from the microphone to the multitrack. Each LCRS module is shipped with an 11 segment LED bargraph meter. The following sections explain the many functions and features of this module.

3.1 Input section

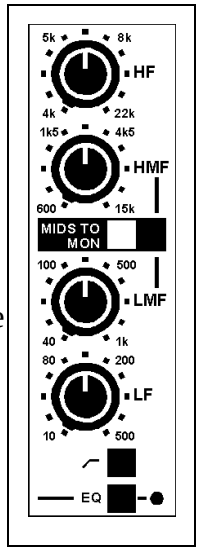
The input section controls all incoming signals such as microphones, keyboards, or any line level device. Fitted at the top of the module is a **48V** phantom power switch for powering condenser microphones or active direct boxes.

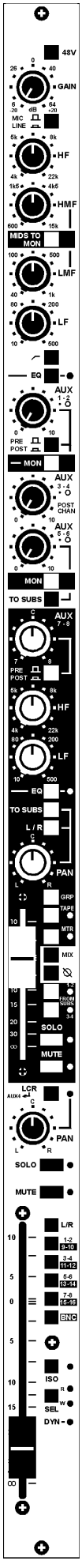


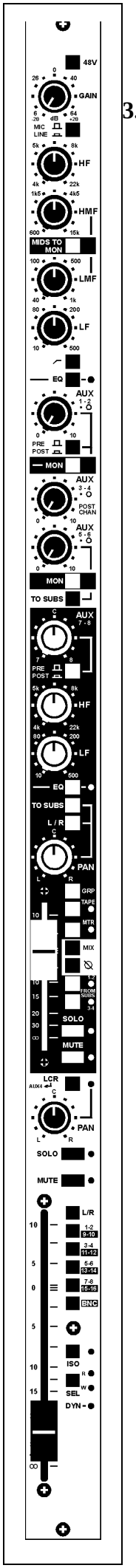
The **GAIN** control is actually one knob which turns two pots adjusting two sets of electronics. If the **MIC/LINE** switch is in the up position, the **GAIN** control will adjust the microphone input and if the **MIC**/**LINE** switch is in the down position, the **GAIN** control will adjust the line input level. In order to make it more accessible, the PHASE switch is located next to the monitor fader. It is used for reversing the polarity (phase) of the microphone *and* line input.

3.2 Equalizer section

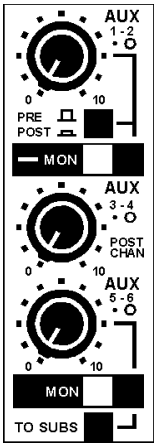
Just below the input section is the four band fully sweepable **Ext**ended Range **"Hi-Def"** (high definition) equalizer. The **HF** (high frequency) band has a boost or cut of 16 dB and is sweepable from 4kHz to 22kHz and is a shelving type. The **HMF** (high mid freq.) band sweeps from 600Hz to 15kHz. A **MIDS TO MON**itor switch allows the two sweepable mids to be moved to the monitor equalizer creating a full four band sweepable equalizer in the monitor path. The **LMF** (low mid frequency) band sweeps from 40Hz to 1kHz. Both mid freq. bands are peak/dip types with a boost or cut of 16dB. The **LF** (low frequency) band starts at 10Hz and sweeps out to 500Hz. It is a shelving type with a boost or cut of 16dB. A high pass filter switch is fitted just below the equalizer to allow the user to cut the low frequencies below 100Hz. It is a 12dB per octave Butterworth type. An **EQ** in/out switch completes the channel equalizer section.







3.3 Auxiliary send section

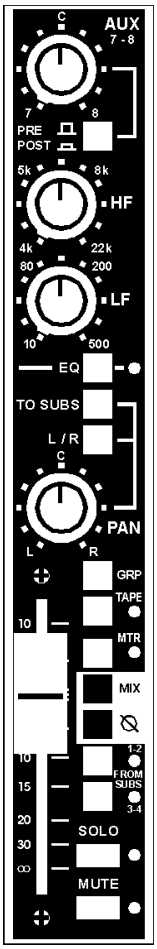


Each LCRS module has 8 discrete aux sends. Aux send 1 & 2 are fed from either the channel path or the monitor path and switchable pre/post fader.

Aux 3 & 4 are always fed post fader from the channel path.

Aux 5 & 6 are post fader fed from channel or monitor. Aux 5-6 sends can be sent to the 16 busses (to subs) to extend the 8 aux sends to 24 aux send busses.
When pressing the Aux 5 & 6 **TO SUBS** switch, Aux 5 & 6 sends will take priority over the monitor and channel bussing.

3.4 Monitor section



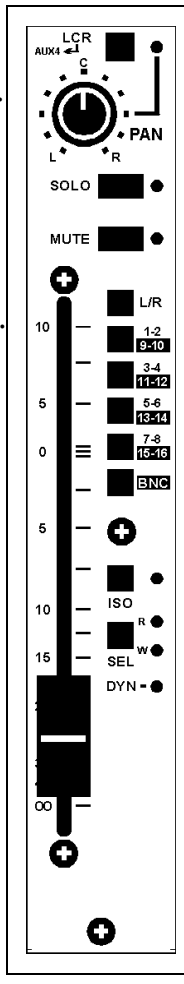
Aux 7 & 8 are always in the monitor path with level and pan control and are pre/post fader switchable.

The **Monitor Equalizer** section is a shelving type with the same extended range as in the channel path. Sweepable from 10Hz to 500Hz and 4kHz to 22kHz in two bands, this **"Ext-Hi-Def"** equalizer has an ultra transparent quality to its sound. The two mid bands of the channel equalizer can be switched into the monitor equalizer making it full four bands.

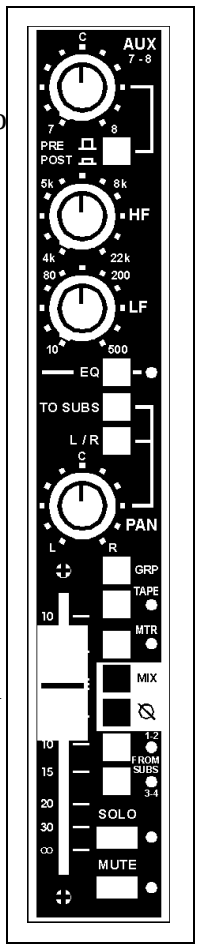
An **EQ** in/out switch allows for easy comparisons.

The monitor section can be assigned to the 16 busses for maximum tracking flexibility. If channels are assigned to the 16 busses and you press the monitor **TO SUBS** switch, the monitor bussing will take priority over the channel bussing.

The **L/R** switch allows the monitor panpot to be assigned to the stereo mix buss. The monitor **PAN**pot pans between the left and right stereo mix busses.
The GROUP (**GRP**) switch inserts the monitor fader into the group output buss in the associated channel and allows the control of a output using the monitor fader as a sub group control.



The **Tape** switch selects the monitor to listen pre or post multitrack. The LED bargraph meter normally follows the tape switch but can be switched (**MTR**) to follow the input signal post channel (balanced) insert.The **MIX** switch swaps the mic / line into the monitor and tape into the channel (from tracking to mixdown mode) on individual modules.



The **Phase** switch reverses the phase for both mic *and* line inputs.

3.5 Floating Subgroup System

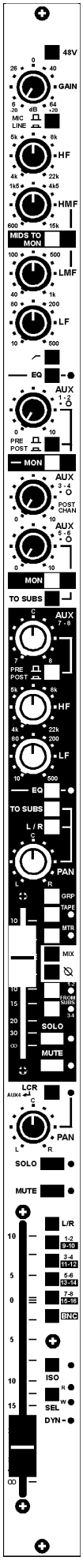
Located in the lower part of the monitor section, the **"Floating Subgroup System"** is one of D&R's most unique features. Although the Orion LCRS has 16 conventional busses, the **FSS** allows the user to buss any input (or any amount of inputs) to those tracks above 16 without having to use the patchbay. As an example, if you have a 32 track tape machine and want to buss some inputs in stereo to tracks 31 & 32, you would assign and pan between busses 1 & 2 (or 3 & 4) on all desired inputs and then go to modules 31 & 32 and press **FROM SUBS** 1 & 2 (or 3 & 4) on both from sub sections. Now a stereo mix of all assigned channels will feed tracks 31 & 32 through the outputs of modules 31 & 32. At the same time, the input sections, eq sections, and channel assign sections of modules 31 & 32 can be used for other inputs.

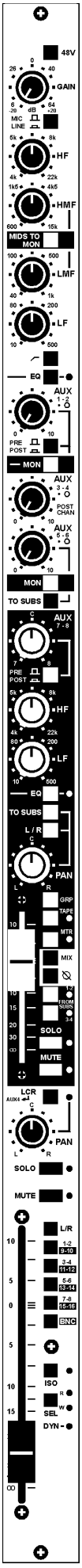
The Monitor section has two **SOLO** modes available, PFL or Solo in Place. The SOLO switch has an LED as an indicator.

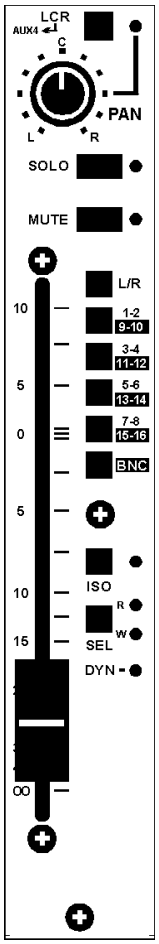
Fitted below the solo switch is an automated **MUTE** switch. **MidiFade** (D&R's midi sequencer controlled automation) and **PowerVCA** (D&R's SMPTE based automation) will control this mute. The **MUTE** switch has an LED indicator.

3.6 Channel assign / fader section

The **LCR** panpot allows for easy surround sound mixes when needed or can be used as a normal panpot when the **LCR** switch is not activated. When active, the center output of the panpot will feed the Aux 4 buss directly creating a third channel needed for the center of Surround mixing. In this condition, the aux 4 send pot (on this channel) is not active.
The surround sound can be fed from aux three if a four channel surround mix is required. If a six channel surround mix is required, aux 5 could be used as left surround and aux 6 for right surround or the stereo aux 7-8 could be used for the left &
right surround sounds. If six channel surround is required, aux 3 would be the sub woofer send.







The channel **SOLO** switch allows for both **PFL** and **Solo in Place** listening and has an LED indicator.

The channel **MUTE** switch fitted below the solo switch is an automated **MUTE**. **MidiFade** (D&R's midi sequencer controlled automation) and **PowerVCA** (D&R's SMPTE based automation) will control this mute. The **MUTE** switch has an LED indicator.

The channel can be assigned to the left / right stereo mix busses by pressing the **L / R** switch.

Assignment to the 16 busses is performed by pressing any or all of the 1 through 8 buss switches. The **BNC** switch bounces between busses 1 through 8 and 9 through 16. The **"Floating Subgroup System"** is routed through the first four busses.

When you press the **ISO** switch on this module, the internal high quality VCA can be individually switched out of the signal path for "audio purists".

The SEL switch changes the automation modes from ISO (VCA in, but not controlled by the computer) **R**ead mode, **W**rite mode, and **U**pdate mode (relative).

An external sequencer can control the internal automation hardware if the optional D&R **PowerFade** automation hardware and software is <u>not</u> purchased.

This module can accept a dynamics option which will be introduced later. The **DYN**amics option will require some hardware and software. Planned for release in late '95, the **DYN**amics option will have compressors, limiters, and gates available.

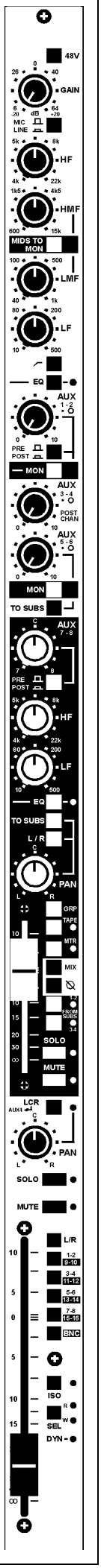
3.7 Automation

The Orion LCRS module comes standard with part of the hardware for **MidiFade**, a MIDI based VCA fader automation or D&R's **PowerVCA** SMPTE based automation system. Channel faders, channel mutes, and monitor mutes are automated in either system with the addition of the optional

LCRS Module....continued

Automation **CPU** with remote and one 8 channel **Multiplex** card for every eight LCRS modules to be automated. If you start with D&R **MidiFade**, D&R **PowerVCA** software and hardware can easily be installed into your 486/66 IBM (or clone) personal computer. After market automation systems can be installed at D&R or in your studio by a certified technician.

Note: As with all console manufacturers and after market automation systems, it is best to have the automation installed by the console manufacturer or automation factory trained technicians.



4.0 The In-line module - description

The Orion in-line module is a basic input / output design whereby all signal flow takes place from the microphone to the multitrack. Each in-line channel is shipped with an 11 segment LED bargraph meter. The following sections explain the many functions and features of the in-line channel.

4.1 Input section

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

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

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

When the **GAIN** control is accurately set, it is possible to achieve the very best signal to noise ratio and maximum headroom required for high quality recordings. This control is for adjusting the line and mic inputs (with seperate elec- tronics), although only one knob adjusts the dual pot.

LINE switches the microphone input to line input on the channel. The line has its own balanced input amp and is controlled by the active (dual) gain control.

MIX, (the record / mix status switch) selects the basic signal path in the module. When in the record mode ("up" position) the mic / line input signal is routed through the channel path with or without equalization and / or aux sends and then to the long fader and channel panpot. It can then be sent to the main Left / Right bus and/or be assigned to the multitrack busses. With the mix switch in the "up" position, the multitrack outputs are heard through the monitor section. When in the mix mode ("down" position) the channel signal (mic / line) is rerouted to the monitor section (light colored), and the tape return input is rerouted to the channel path.

The **TAPE** switch allows you to choose from where the monitor gets its signal. In the up position (source), the channel fader feeds the monitor through summing amps 1-8. For channel selections above 8, the channel fader will feed the monitor directly. In the down position (tape), the tape return (monitor in) feeds the monitor signal path.

4.2 Equalizer Section

This four band semi parametric equaliser 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. The two mid bands are sweepable with a boost or cut of 16 dB and are bell curve characteristics. The HMF (High / Mid Frequency) and LMF (Low / Mid Frequency) can be switched into the monitor path.

The **100 Hz Highpass filter** is a fixed 9 dB per octave Butterworth model which reduces low frequency noise effectively and musically. It can be switched on or off in the channel path.

The **HF** (high frequency) section is a varible frequency shelving type, sweepable from 4,000 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 with varible frequency ranges from 600 Hz to 15,000 Hz and has a maximum boost or cut of 16 dB. The bandwidth has a Q factor of 1.5.

The **LMF** (Low / Mid Freq.) section has level and frequency controls with varible frequency ranges from 40 Hz to 900 Hz and has a maximum boost or cut of 16 dB. The bandwidth has a Q factor of 1.5.

The HMF and LMF bands can be switched in or out of the channel and / or monitor circuit paths by pressing the MF to MON switch. If the equalizer is only partly inserted in a signal path, the equlizer (EQ) switch has priority over the monitor switch. 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 (and noise). Test comparisons of other equalizers helped the D&R design team to create an equalizer that sounds very musical , but at the same time, raising the standard in specs and sound quality. Noise and distortion are kept to an absolute minimum.

4.3 Auxilliary Send Section

The Orion has eight auxilliary send busses. While in the tracking session, some aux sends are used for feeding the headphones and some are used for effects sends. During the mix session, all aux sends are used for effects sends.

Auxilliary sends **1&2** are on dual concentric controls. The top control is the send control for aux 1 and the bottom control is the send for aux 2. These Aux busses are normally used for stereo headphone sends in the tracking session.

Aux 3&4 are also on dual concentric controls. The top control is Aux 3 and the bottom control is Aux 4.

Aux sends 1 - 4 can be fed from either the channel path or MONitor path and are PRE / post switchable.

Auxilliary **5-6** are switchable to **Aux 7-8** busses.

4.4 Floating Subgroup System

The **"Floating Subgroup System"** is one of D&Rs most unique features. Although the Orion has 8 conventional busses, the FSS allows the user to buss any input (or amount of inputs) to those tracks above 8 without having to use the patchbay. As an example, if you have a 32 track tape machine and want to buss some inputs in stereo to tracks 31 & 32, you would assign and pan between busses 1 / 2 (or 3 / 4) on all desired inputs and go to modules 31 & 32 and press FROM SUB 1-2 (or 3-4) on each from sub section.

Now a stereo mix of all assigned channels will feed tracks 31 & 32. The **"Floating Subgroup System"** can be used for 48 and 64 tracks as well. The Orion series bussing is only limited by the amount of input / output channels in any given frame. The FSS is one of the most misunderstood features, however, once understood, it can be the most used and appreciated feature in the Orion.

4.5 Monitor Section

The monitor section is the second signal path in the Orion in-line channel. It has its own volume control, panpot, and mute & solo switches as well as a Group switch and fader reverse switch.

In record mode (the mix switch in the up position), the monitor section is fed by either the tape return or group output (tape return or tape send). In mix mode (mix switch depressed), the monitor section is fed by the mic / line input and the tape return now moves to the channel and is controlled by the long fader.

Two mid bands of the equalizer can be inserted into the monitor section. This feature makes a fully functioning input usable in the mix mode for signals other than the channel signal. Channel and monitor paths can have either tape return or mic / line inputs with a two band equalizer (on each) at the same time.

The **GRP** (group) switch inserts the monitor pot into the output summing amps of that particular channel. All relevant functions of the monitor section are also inserted. The stereo solo-in-place or pfl (pre fade listen) solo system and mute can be used on this group fader as well as auxilliary sends and equalizer, but control of the monitor signal path is obviously lost, however is still monitored. The mix channel status switch has no infuluence on the group switch.

The monitor pot and channel fader can be reversed (REV) when use of the fader for monitoring during recording is preferable. All functions such as assigned Aux sends, solo, and mute functions stay in their normal signal path. Only the fader and monitor pot are electronically reversed.

The **SOLO** switch has two modes, pfl (pre fade listen) or a "destructive" stereo solo-in-place system. Master status
switching (located in the master section) selects the status of "solo- in-place" or "PFL" mode for the entire console. When activating any solo switch in the pfl mode, it will send the prefader signal of the monitor section to the CRM speakers in mono. In the solo in-place mode, the post monitor panpot signal is heard (where panned in stereo), and all other channels are muted within the stereo mix buss. A solo indicator LED is fitted next to the solo switch.

The **MUTE** system is designed around a special soft-muting integrated circuit, completely click-free. A mute indicator LED is fitted next to the mute switch.

NOTE: Always make sure that unused monitor sections are muted as the unterminated inputs will degrade the excellent low noise performance of the mixing amps.

4.6 Channel assign / Fader section

The Channel assign section allows the user to assign to any of the 8 busses as well as the "Floating Subgroup System". The routing switches buss to the 8 multitrack output busses using four pushbuttons. With each switch routing to a pair of summing amps, you have the choice of assigning to the odd or even amp using the channel panpot.

The **MIX** switch (assign section) assigns the channel panpot to the stereo mix buss while the channel pan control pans the signal between the main stereo mix **L / R** busses and / or the odd and even multitrack summing busses if assigned.

The summing amp and internal structure yields the Orion's extremely quiet and distortion free sound, therefore a direct button to bypass the amps is not necessary. Every channel is individually assigned to the multitrack input which allows for tracking to be simple and consistant.

The **SOLO** switch has two modes, pfl (pre fade listen) or a "destructive" stereo solo-in-place 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 channel section to the CRM speakers in mono. In the solo in-place mode, the post channel panpot signal is heard where panned and in stereo and all other channels are muted within the stereo mix buss. A solo indicator LED is fitted next to the solo switch. The solo LED has two functions; when in the solo mode it is a solo indicator and when that channel is not soloed, the LED is a post EQ peak indicator for the channel input.

The **MUTE** system is designed around a special soft-muting integrated circuit, completely click-free. A mute indicator LED is fitted next to the mute switch.

NOTE: Always make sure that unused channel sections are muted as the unterminated inputs will degrade the excellent low noise performance of the mixing amps.

Fader section

The Orion has a high quality 100mm fader. Alps or P&G faders are optional. You can order either the PowerVCA Automation (D&R's VCA fader & mute automation), PowerFade (D&R's moving fader automation), V/Desk, Uptown, Tetra, or any other after market automation system. Any of the above automation systems can be built-in at the factory or installed later. If you choose to add automation later, it is always best to talk to your D&R representative first.

4.7 LCRS/In-line Module connectors

Input and Output connectors.

In addition to the optional patchbay, every channel also has the following connectors at the back of the housing. The in-line module backplates are double width to accomadate two channels.

The balanced Group (GRP.) output is a stereo 1/4" jack socket:
Tip = hot
Ring = cold
Sleeve = shield

The MONitor insert is on a stereo jack socket.
Tip = return
Ring = send
Sleeve = shield

The CHANnel insert is a 1/4" stereo jack socket:
Tip = return
Ring = send
Sleeve = shield

The balanced MONitor input is a stereo 1/4" jack socket:
Tip = hot
Ring = cold
Sleeve = ground

Note: The default setting on the monitor inputs and group outputs are +4 dBu. You can choose jumper settings of -10 dBV on the channel (PCB) boards using small needle nose pliers.

The balanced LINE input is on a 1/4" stereo jack socket:
Tip = hot
Ring = cold
Sleeve = shield

A balanced XLR connector is for the MICrophone input.
Pin 1 = shield
Pin 2 = hot (in phase)
Pin 3 = cold (out of phase)

5.0 The Dual Stereo Aux Return module

This stereo return module is one of the most flexible stereo modules ever designed. It actually has two stereo returns (four inputs). Although designed for effects returns, this module can be used for stereo keyboards, drum machines, or any device needing two inputs on one fader. The paragraphs below describe stereo return A in full. Stereo return B is identical.

5.1 Input section

The input section has a stereo GAIN pot controlled by one knob. The gain control is a dual pot used to adjust the gain of two line amps (stereo input). The adjustment range is from -20dB to +20 dB. The backpanel jack sockets actively mixes the two inputs into a mono signal when only one input is used. This is done when you need to feed the output of a mono device to both channels.

5.2 Equalizer section

The two 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. The low frequency band is a shelving type at 60 Hz. Each band has a boost or cut of 16dB for both left / right channels.

5.3 Aux send section

The Aux send section has dual concentric pots to feed Aux busses 1&2 which is used for stereo effects returns to the stereo phone system as well as sends to other devices. Aux sends 1&2 are pre / post switchable. Aux send 3 (5) and 4 (6) send a post summed mono signal to the master aux busses. Aux 3 & 4 can be switched to Aux busses 5 & 6.

5.4 Floating Subgroup System

The stereo return can be bussed to any pair of outputs on in-line modules which allows for routing to any multitrack inputs. Also fitted on each stereo return section is a stereo balance control (*not a pan pot*), solo switch with indicator, mute switch with indicator, and 60mm stereo fader. The lower section "B" is identical to the upper section "A".

5.6 Dual Stereo Aux Return Module Inputs

Input connectors.

The Dual Stereo Return modules are not wired to the patchbay inside the console because some owners would rather have their reverbs go to dedicated inputs in order to free-up patch points for more signal processing. If you would prefer having the Dual Stereo Returns in the patchbay, we would suggest wiring them to the last rows in the far right column of the tie lines. A permanent magic marker can color the blue jacks black (all inputs). If you need assiatance, call your D&R Representitive.

The balanced inputs are on 1/4" stereo jacks.

Tip = hot
Ring = cold
Sleeve = shield

There are four input jack sockets on the back panel of each Dual Stereo Return module

6.0 Optional Patchbay - description

The patchbay is designed around Bantam type tiny telephone jack sockets. This optional patchbay is completely modular and can be expanded as your budget allows or ordered complete. All master inputs / outputs and 120 tie lines (for signal processing) are standard when you order the patchbay. The entire patchbay is balanced and internally "starground" wired.

6.1 Patchbay - points

Channel patch points from left to right are:

Line input - channel insert send & return - monitor input - monitor insert send & return - group output.

Each four rows of channel patch points are followed by four channels of multitrack inputs and outputs. The tape inputs and outputs are normalled to group outputs and monitor inputs inside the patchbay.

The **Master Section** has six rows of Bantam type jacks.

First Row: Left master insert send & return, right master insert send & return, +4 outputs left & right, -10 outputs left & right.
Second Row: Tape or mastering machines A, B, & C inputs left & right.
Third Row: Tape or mastering machines A, B, & C outputs left & right.
Fourth Row: Tape, DAT, CD, or two track machine playback inputs to CRM.
Fifth Row: Aux send outputs 1 thru 8. You can normal these jacks to the inputs of selected signal processing.
Sixth Row: Two sets of four mults (parrallels) used for patching the output of one device to the inputs of other devices.

Tie Lines: The tie lines (120 in total) are in rows of eight. The tie lines are for any signal processing equipment. For ease of use, all outputs are on blue sockets, and all inputs are on black sockets. The drawing shows the first 40 (of 120) tie lines. Interfacing with external machines, effects, or power amplifiers can be performed via the connector panels on the backplates of the console or via the 25 pin sub 'D" connectors on the patchbay connector panel.

7.0 Instructions for operation

The Orion is designed to be the perfect recording /mixing console for multitrack, post production, and MIDI studios. In order to get more familiar with the Orion, we shall discuss the entire recording process and divide it into five basic situations. Situation 1 through 4 are for the more conventional recording studios, and situation 5 is for the MIDI studio.

1. The session - Recording from microphone or line input to the multitrack machine. This could be from one or more channels at a time.

2. The playback - In this mode you would listen to what has been recorded on the multitrack machine.

3. The overdub - Overdubbing is listening to already recorded tracks and recording on empty tracks until all tracks are filled.

4. The remix - Playing of all recorded tracks together with signal processing equipment and all that is necessary to create the final mixdown.

5. The MIDI or Virtual Session - Programmed keyboards, drum machines, reverbs, effects, vocal and/or instrument tracks from a multitrack machine and who knows what else, all at the same time.....direct to your Dat Machine, two track master machine, Hard Disk editing system, CD recorder, or cassette deck.

7.1 Situation 1 - The Tracking Session

Recording basic tracks - 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 channel signal. The signal flows through the fader and is available postfader to be routed by way of the assignment switches which can feed the input to your multitrack recorder. The LED bargraph reads the outgoing signal.

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 the mike input only.

After plugging in a mike or line signal, depress the channel solo switch above the channel fader you are setting, set the solo status switch to pfl in the master section, then turn the gain control (of that channel) clockwise until a "0" output level is reached on the master meters. Now slide up the channel fader to "0". Now switch the solo out. If the signal source 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.

Monitoring

With the Orion series, you are able to monitor your multitrack by way of the separate monitor section. The monitor section of the in-line module allows you to have two usable inputs, both with EQ, both being able to send to the aux. busses, both with their own volume control, panpots, mutes and solos, and able to be routed at the same time.

Multiple Modules Assigned to One or Two Tracks

When more than one microphone or line signal has to be recorded on a single track or in stereo on two tracks, a submix facility is required. This can be done easily on the Orion by way of the the internal subgroup amplifiers located on every channel module or the four "Floating Subgroups". Simply route to one of the 8 subgroups by activating a channel routing switch on as many input modules as required. Decide on which track you wish to record these signals and activate the related number. The channel metering will show the subgroup level which can be changed overall by the monitor group fader (Grp switch must be pressed). In order to monitor these tracks on the modules, the tape switch should be in the up position for monitoring pre-tape (console out) and in the down position for monitoring post-tape (tape switch down). If you wish to route several inputs to a track higher than 8, you would use the "Floating Subgroup System" (see section 3.4).

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 even in the monitor insert if an entire group signal is to be processed. To do this, it is necessary to activate the GRP switch in the summing channel to use the monitor insert.

Headphone (Que)

During recording it is essential that the talent hear an independent mix of what the engineer and producer are hearing. Headphone mixes are usually derived from pre-fader auxilliaries. In the Orion aux 1 & 2 are ideal for this purpose. The best way to build a mix for the headphones is to have the monitor section of the in-line module feed aux busses 1 & 2. When there is limited time to set up a headphone mix, give the talent the CRM mix (in the Phones section of the master modules) and build up an independant headphone mix on aux 1 & 2 when time allows.

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 the modern recording or MIDI studios of today, there is a demand for many effect returns and inputs for MIDI related gear. For that reason D&R has designed the Orion with stereo effects return modules. See section 5.0 of this manual for a complete description of this module. Any unused channel or monitor input can also be used for returning effects. Every channel can accept two returns with equalization and aux send capabilities. Due to the phase uncertainties of stereo reverbs and such, it is always best to keep anything in stereo in the stereo domain. You will find that the dual stereo return module has the lowest cost per input ratio of any available modules in the Orion series.

7.2 Situation 2 - The Playback

Multitrack playback

The Orion gives you a convenient way of monitoring your multitrack recorder. Put all the tape switches in their down position. Now the tape outputs are feeding the monitor path and you can adjust the amount of signal you desire and pan it within the stereo image. Most multitrack recorders today have input monitoring on their remote which will allow for monitoring tape until you press record and then it will monitor what you are sending to the track.

Auxilliary sends can be switched to get their signal from either tape or channel and equalization can be inserted in both signal paths whenever needed. Independent control over processing is monitored by solo in either signal path.

7.3 Situation 3 - The Overdub

Multitrack synchronizing

Overdubbing is the process of building up a recording track by track while listening to previously recorded tracks.

The Orion has an in-line monitor for each track of the recorder making it easy to overdub. Connected to the monitor section of the in-line module, you push all tape switches down (located above the equalizer) and do all your sync switching from the tape machine or remote. The headphone mix is on the aux send 1 & 2 busses. Aux 1 & 2 should get their signal from the monitor section. It is best to activate aux 1 - 2 or 7 - 8 PRE fader switches anytime you're using Aux 1 & 2 or 7 & 8 for a headphone mix (depending on whether you have standard or LCRS modules.

7.4 Situation 4 - The Remix

Multitrack mixing

Remix is the process of combining all recorded tracks with (keyboards and drum machines for MIDI) signal processing and sending the mix to a two track reel to reel master machine, Hard Disk editing system, CD recorder, DAT machine, or cassette recorder. On the in-line module you must push the upper "mix" switch down. This routes the tape return to the channel input and routes the mike/line inputs to the monitor section of the module. At this point you can use either a mic or line input in the monitor section which will feed the stereo mix buss. This will give you two inputs per module in the final mix. You can activate the **MF to MON** switch and have the **HMF** and **LMF** sections of the equalizer on the monitor section while the **HF** and **LF** bands remain on the channel.

You must activate the **EQ** switch if you desire EQ on the channel or monitor. The incoming signals can be routed to the stereo mix buss via the **"MIX"** switch in the channel assign
section. The monitor section normally feeds the stereo mix buss. Sub groups can be made up (as required) in the same way as during recording. Aux sends 1 - 4 can get their signal from the monitor section or channel path with pre / post switching. Aux sends 5 thru 8 get their signal from the channel section in the post fader position.

7.5 Situation 5: MIDI or The Virtual Session

Virtual Tracks: The MIDI Set-Up

In most MIDI studios there will be an eight-track rather than a sixteen or twenty four track tape machine. In the MIDI studio, the majority of music production is programmed on a sequencer using MIDI keyboards, sound modules, drum machines, or other MIDI related equipment.

Therefore, you will only require tape tracks for vocals and those instruments not adequately reproduced on today's keyboards. If there is a multitrack recorder in the MIDI studio, one of the tracks would be used to record time code (SMPTE or MIDI code). This will allow your sequencer to keep keyboards, drum machines, and other MIDI equipment synchronized.

The Orion was designed with the multi-track and MIDI studio in mind. In today's medium to large MIDI studio, there is a need for as many as 100 inputs to be used for everything from tape tracks to keyboards and drum machines. For this reason, the Orion, when fitted with both in-line modules and stereo return modules, can net over 100 inputs in the virtual track session or mix down.

8.0 Installation - Electrical

8.1 Local Electrical Voltage

Before connecting the Orion, 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 is a 6.3 amp fuse with a 125 volt rating (10 amp fuse in America with a 250 volt rating). After replacing a blown fuse with the correct size and rating, turn the power supply 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.

NOTE: DO NOT REPLACE THE FUSE WITH ANY OTHER TYPE AS THIS CAN BECOME A SAFETY HAZARD AND WILL VOID YOUR WARANTY.

8.2 Electrical Wiring

To take full advantage of the excellent signal to noise ratio of the Orion, 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. All equipment using three wire ac connectors should have a ground lift adaptor on each cable before plugging into the ac outlet. In most cases, the incoming electrical ground is inadequate and a dedicated ground system should 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 ground rules to follow. All signals in a recording studio are referenced to ground. This ground must be clean and free of noise. A central place (central to <u>all</u> equipment) should be selected as the "central star ground point" and all grounds should terminate at this point. This point can be a solid metal plate with at least 50 places to hookup all incoming grounds. This is commonly referred to as a "star ground system".

In some instances electrical contractors will daisy chain ground connections in the ac distribution system. This is <u>not</u> suitable for a studio. Ideally, run a separate ground wire from each piece of equipment to the "central star ground point". The "central star ground point" should be connected to a pair of eight foot ground rods using larger (#10) wire than your equipment ground wires. Separate and identify "clean" and "dirty" AC outlets.

Use clean outlets for audio equipment and the dirty ones for lighting, air conditioning, cola machines etc. Do not intermix these two types of outlets. AC interferance can be greatly reduced by using an isolation transformer or some type of balanced ac power device to power outlets. Ground this transformer directly to the "central star point". After all equipment is connected to the ac power, check with a ohm meter or continuity tester to be sure of no possible chance of ground loops.

All equipment should be physically located as far as possible from the main breaker panel and should be totally isolated from the equipment rack and other equipment so ground loops are avoided. Equipment can be mounted in wood rack rails to avoid ground loops.

Now you can run a #12 stranded wire with jacket from each piece of equipment to the "central star ground point". All ground wires should be the same length with a tolerance of plus or minus 10% in order to have the same ground potential everywhere. On the equipment ends of each ground wire you should solder a round hole screw terminal. Remove a chassis screw from each piece of equipment and file the paint in that area so it will make good contact when you connect the terminal. Next, connect the ground wire termanal to each piece of equipment and connect each wire at the other ends to the "central star ground point".

9.0 Installation - audio

9.1 Interface CRM Levels

The Orion in its standard configuration can interface with all available equipment. As a service of D&R, free installation consultation is available. Contact your supplier for details.

Attention concerning the CRM 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.

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

9.2 The Initial Hook-Up

First connect the rack-mounted power supply to the console. All faders, monitors, 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.

a. Connect the CRM 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 interference. Slowly turn the CRM 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. After checking the main power amp and speakers, check CRM 2.

b. Before making any other connections move each monitor fader to the 0 dB position with the tape switch depressed on each monitor section. Connect the multitrack outputs to the tape return connectors on the backplate of each in-line module (or via the 25 pole sub-D connector if the optional patchbay is installed), then connect each connector on the tape output of your multitrack. Check for hum or noise after each track has been hooked up. "Hiss" will normally increase slightly with each track. Connect the tape send output jacks to the inputs of the multitrack. Carefully listen for excessive noise or hum. If after hooking up an input or output excessive noise or hum is detected, stop and take corrective action before proceeding. Do not hook up all 16, 24, 32, or 48 tracks and then listen. You may need to rewire the entire cable harness to make the system clean.

c. Connect stereo tape recorders (inputs and outputs), stereo headphone amp, and all signal processors.

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

9.3 Shields & Grounds of 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 connect the other end of the shield through a 0.01 microFarad capacitor. This will ground the RF but will not affect audio frequencies. 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 paragraph above. Remember, the shield is not considered to be "ground" and it should only be connected at the output of any device. There are only a couple of exceptions to this rule, one is patch cords and the other is microphone cables. We realize that the correct interfacing of 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 to be semi-professional. Because many semi-professional tape machines are built to professional specifications, D&R builds into the Orion console the ability to interface with both levels.

Note: When checking your new Orion console for noise, you will notice that the console is extremely quiet without any external equipment hooked up. D&R is not responcible for the noise you will experience when interfacing other equipment. Since you are use to other consoles when first encountering a D&R, the lack of noise from your Orion makes you more aware of noise from other equipment as you hook it up. D&R recomends using the highest quality external equipment with the Orion. Because high quality sound must be monitored with speakers and amps with extreme specs, D&R suggests using Pacific Innovative Amps to drive your speakers. Ask your D&R representative for more information.

10.0 Connecting the Orion

10.1 Master module # 1 (CRM & Solo)

Master insert left	Tip = return Ring = send Sleeve = ground
Two track A, B, & C	Tip = hot Ring = cold Sleeve = shield
-10 dBv left output	Tip = hot Ring = n / c Sleeve = ground
+4 dB left balanced output	Tip = hot Ring = cold Sleeve = shield
CRM 1 & 2	Tip = left Ring = right Sleeve = ground

10.2 Master module # 2 (Oscillator and Communications)

Master insert right	Tip = return Ring = send Sleeve = ground
Two track A, B, & C Right (playback)	Tip = hot Ring = cold Sleeve = shield
-10 dBv left output	Tip = hot Ring = n / c Sleeve = ground
+4 dB left balanced output	Tip = hot Ring = cold Sleeve = shield
Power supply	Pin 1 = Ground Pin 2 = +18 vdc Pin 3 = -18 vdc Pin 4 = +48 vdc Pin 5 = Ground

10.3 Master module # 3 (Aux 1, 3, 5, & 7 and Studio)

Aux 1 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 3 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 5 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 7 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Studio Output	Tip = left Ring = right Sleeve = ground

10.4 Master module # 4 (Aux 2, 4, 6, & 8 and Phones)

Aux 2 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 4 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 6 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Aux 8 output (balanced)	Tip = hot Ring = cold Sleeve = shield
Phones Output	Tip = left Ring = right Sleeve = ground

This concludes the master section module connections. Refer to section 8.0 through 9.4 of this manual for usefull ideas and wiring techniques.

10.5 Connecting the In-Line / LCRS Module

Description:	Connector:	Connection:
Group output (tape send) balanced	1/4" TRS	Tip = hot Ring = cold Sleeve = shield
Monitor INSert send & return	1/4" TRS	Tip = return Ring = send Sleeve = ground
Channel INSert	Same as above	
MONitor input (tape return) balanced	1/4" TRS	Tip = hot Ring = cold Sleeve = shield
LINE input balanced	1/4" TRS	Tip = hot Ring = cold Sleeve = shield
Mic input (balanced)	XLR	Pin 1 = shield Pin 2 = hot + Pin 3 = comm -

11.0 Troubleshooting and servicing

11.1 Troubleshooting

It is essential to study the signal flow chart carefully, Knowing and understanding the signal flow of your console will help to isolate problems. By tracing the signal from input to output jacks, you can locate and correct the 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.

11.2 Removing a Module

The Orion 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 LEDbar, master section and backplates. 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. If servicing the LEDbar meter, remove the backplates behind the LEDbar section which hold the input / output connectors. It is often easier to also remove the backplates positioned left and right of the module under test. Unplug the ribbon cable from the ledbar and remove the upper and lower screws from the meter front panel and you are ready to remove the meter. If servicing the module, remove the
metal strip underneath the LEDbar front which holds the plexiglas in place and conceals the screws retaining the module. It is now possible to remove the two module retaining
screws and carefully lift the module until all flat cable connectors on the bottom of the module PCB can be unplugged. 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 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.

Note: Turn the power supply off each time you pull or replace a module!

11.3 Patchbay - servicing

The patchbay is fully modular and can be serviced after first removing the mounting screws, then lift up on the (armrest) end of the patchbay. A one foot stick can be used to prop up the patchbay while servicing it. Remove the cables attached to the card that needs servicing. The card can be removed after unscrewing two screws and then push the patchpanel card downwards. The card will still be connected to the star ground system which will need to be unconnected before the card can be removed from the console.

SPECIFICATIONS:

Inputs:
Mic input, balanced, RF suppressed, 2 kOhm.
C.M.R.R. at 50 Hz, -70 dB.
Sensitivity: -80 dBu max for +4 dBu output.
Noise mic: -129.5 dBu, (A-weighted) 150 Ohms.
Line inputs: balanced, 10kOhms, -20dB to +20dB.
Tape inputs: +4 dBu balanced / -lOdBv unbalanced.
Mon./chan. lnserts: 10kOhm/47 0hrn unbalanced OdBu.
Stereo machines: +4 dBu balanced -10dBv unbalanced

Outputs:
Main outputs, + 4dBu balanced / -10 OdBv unbalanced.
Tape outputs: + 4dBu / -10dBv balanced.
All other outputs + 4dBu balanced.
Noise master fader down: -lOO dBr (A-weighted).
Noise 32 channels routed: -87 dBr (A-weighted).
Noise 64 channels routed: -84dBr (A-weighted).

Equalization:
High pass filter, -3dB at 100Hz.
H.F. +/- 16dB from 4kHz to 20kHz (22kHz) shelving.
L.F. +/- 16dB from 20Hz (10Hz) to 500Hz shelving.
H.M.F. +/- 16dB from 600Hz to 15kHz bell, Q factor 1.5
L.M.F. +/- 16dB from 40 Hz to 900Hz bell, Q factor 1.5
H.M.F. and L.M.F. switchable between channel and monitor with bell curves
High pass in channel path only

Overall:
Nominal internal operating level OdBu (0.775 V).
Frequency response, any input to any output anywhere in the console.
20Hz-20kHz,-0.5dB.10Hz • 100kHz, -3dB.
Total harmonic distortion:
Mic. in Grp. out: 1 kHz: 0.015%, 10kHz: 0.025%.
Maximum output: +26dBu into 600 Ohm balanced.
Maximum headroom: not less than +22dB

Crosstalk:
Mic to line: < 105 dB at 1 kHz.
Channei mute: < 110 dB at 1kHz.
Pan-pot isolation: < 75dB at 1kHz.
Channel routing: < 9OdB at 1kHz.
Channel fader: < 95dB at 1 kHz.
Aux send kill: -9OdB at 1 kHz

Standard Configuations:
Frarne 38 (32 in-lines and 12 stereo returns).
Frame 51 (32 in-lines and 12 stereo returns with patchbay).

Note: Patchbay 32 width equivalent to 13 module positions.

Weight: 30 frame: 60kg / (132 Lbs).
38 frame: 90kg / (198 Lbs).
51 frame: with patchbay 100kg (220 Ibs).

Dimensions:

Height: 966mm / 38.03"
Depth: 929.68mm / 36.60"
Width:
Orion 30 chassis: 1282 mm / 50.47"
Orion 38 chassis: 1561 mm / 61.46"
Orion 51 chassis: 1995 mm / 78.54"

Options:
Conductive plaslic faders
Analog VU Meters (master)
Pedastal base
MidiFade Automation
PowerMute Automation
PowerVCA Automation
PowerFade Moving Fader Automation
After-market automation: V/Desk - Uptown - Tetra

D&R reserves the right to change specifications without prior notice. As with all D&R products, the Orion exceeds most of the printed specifications.

Factory / Head office:

D&R Electronica b.v.
Rijnkade 15B
1382 GS WEESP
The Netherlands

Phone: (--) 31 2940-18014
Fax: (--) 31 2940-16987
Email: info@d-r.nl
Web: d-r.nl

DECLARATION OF CONFORMITY

Manufacturers Name: D&R Electronica Weesp b.v.

Manufacturers Address: Rijnkade 15B,
1382 GS Weesp,
The Netherlands

declares that the product

ORION / ORION-X

conforms to the following product specifications:

EMC: **EN 55022: 1987**
CISPR 22 (1993) class B

EN 500082-1 (1992)

Supplementary Information:

The products herewith complies with the requirements of the EMC Directive 89/336/EEC (1989) as amended by the CE Marking Directive 93/68/EEC (1993).

D&R Electronica Weesp b.v.
Rijnkade 15 B
1382 GS WEESP
The Netherlands
President of Engineering