"4000 SERIES"

USER MANUAL

D&R

INTRODUCTION

The D & R 4000 series is distinguishable from other desks, employing in-line mixing in various ways. Firstly a word about how we've achieved such technical excellence for such a competitive price.
We have employed a plug and flatcable wiring system which means that the individual channels are interchangeable with each other thus making the desk fully modular. Another technical innovation which helps hold the price of the 4000 series down is a completely new way of routing which eliminates the need for the complicated switching and wiring circuits found on conventional desks leading to a saving of 150 switches on a 32 channel mixer.
A simple patch-bay which facilitates very easy patching is also the result of our cost conscious design efforts.

The console itself is of a superb design, the modules are mounted individually in aluminium "U" profiles with clear unerasable lettering protected by a polycarbonate film. These profiles are set into the sturdy metal housing which has attractive wooden sides.
The well designed control lay-out facilitates pleasant mixing without the confusion that usually arises from the array of knobs and switches found on mixing desks in general.
The mixers in the 4000 series rank among our finest products and are the result of 10 years research into mixing desk design which has led to many innovations and developments enabling us to offer you superb desks at very competative prices.

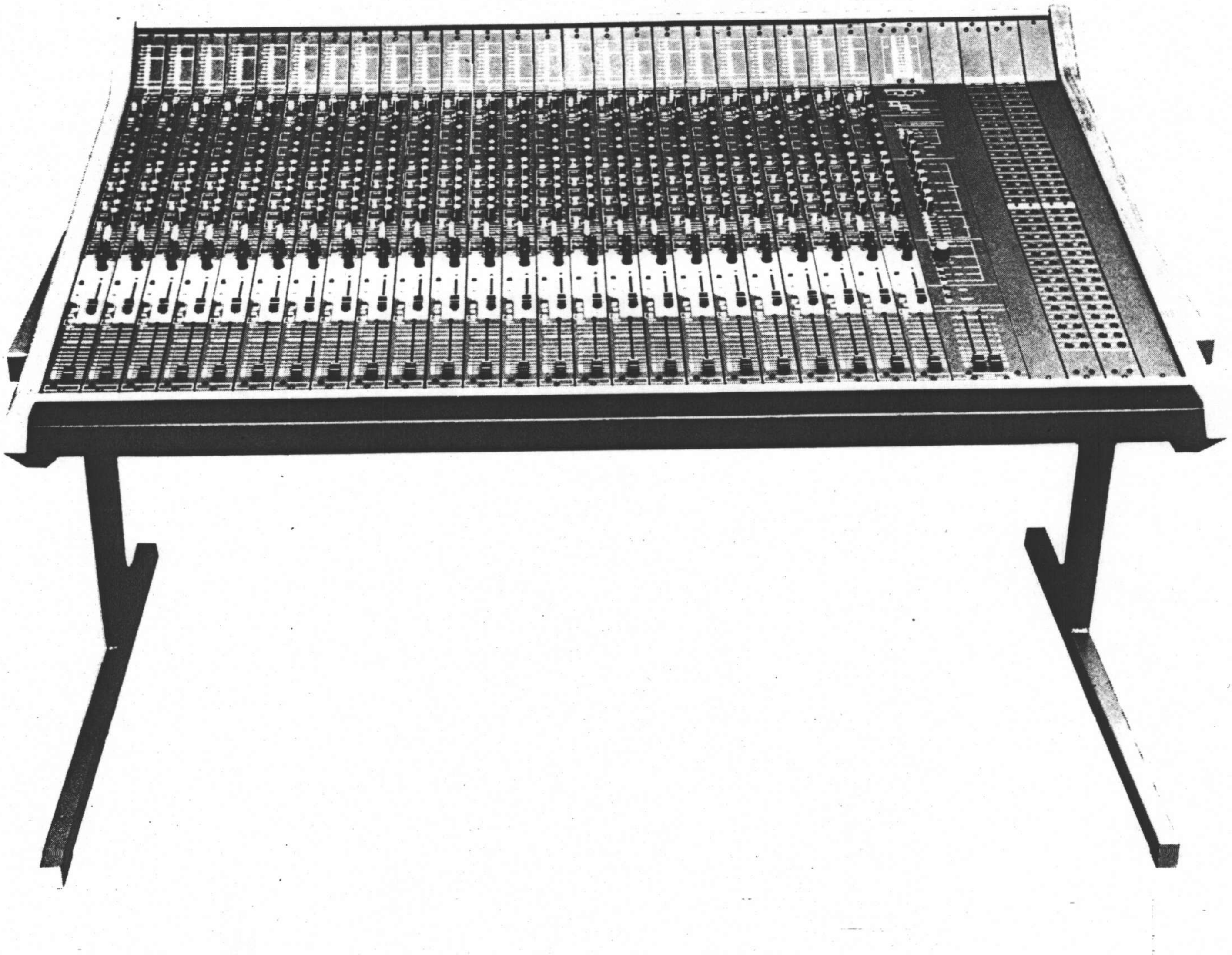
Design Highlights

F.E.T. switching is employed giving practically unmeasurable low distortion and a shut off in excess of 90 dB.
A completely new approach to limiting of above audio range frequencies, through passive filtering (instead of the standard active filtering) gives this console as all our other designs an incredible transparancy through its absence of transient distortion. By critically damping every integrated circuit at 40 KHz square waves we have achieved complete elimination of overshoot and/or ringing and slewing.
At the mic inputs all the amplification is performed by discrete low noise transistors and throughout the signal path by Bi-Fet op-amps (series TL 070), while the mixing amps utilise the reknowned industrial standard low noise audio op-amp NE 5534 AN.

We have chosen for a minimum audiopath to achieve total transparancy.
Due to an excellent circuit design there is a minimum of crosstalk, control interaction and this combined with the superb printed circuit board lay-out contributes to a very stable and low noise product.

Power supply

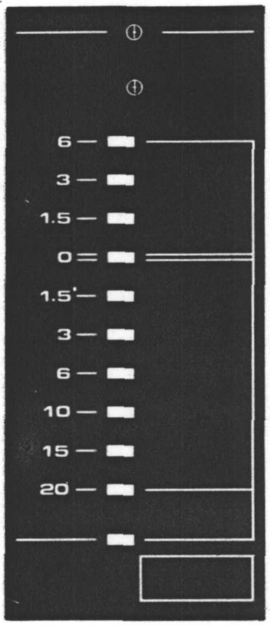
The power supply is housed in a 19", rack mounting case and consists of three fully independant and protected powersupplies with very low ripples.



A short rundown of the series 4000 possibilities

- 11 segment positive/negative reading peak-bargraph meter per in/output channel
- 48V phantom powering, switchable per channel
- click free phase reverse switch for mic/line and remix signals entering the console
- extremely low noise electronically R.F. screened, balanced mic amps
- simultaneous sync/remix inputs for + 4 dBu as well as – 10 dBV
- 100 Hz high-pass filter
- 4 band sweep eq. of novel design without interacting control functions
- 6 aux sends pre/post switchable and selectable from channel and monitor signal paths
- completely new free floating in/outputs from subgroup amps making subgrouping to any multitrack channel possible with a minimum of switches
- simultaneous routing to master, direct output and group summing amps possible
- sync and effect inputs per channel, changing a 24 console into a 48 line input remix console
- monitor mute and p.f.l.
- channel mute and p.f.l.
- 100 mm channel fader, 60 mm monitor fader
- 2 inserts per channel
- simultaneous multitrack feed outputs for + 4 dBu and – 10 dBV available
- master section with 25 segment led bars and phase correlation meter
- low distortion 1 KHz line up oscillator
- talkback with built in electret and routing
- communication system switchable via the a.f.l./p.f.l. system
- 6 master aux sends with individual selectable a.f.l. switches
- comprehensive control room monitor section with alternative monitor loudspeaker switching, mono switch and mute switching
- two stereo master recorders can be played back (+ 4 dBu/ – 10 dBV inputs available)
- modular 64 point patchbay modules
- all connections via XLR and jackplugs

DESCRIPTION OF CONSOLE CONFIGURATION

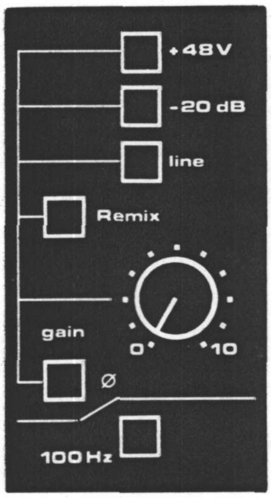


Ledbar

The 11 segment ledbar (the first led indicates *only that the power supply is on)* is a peakreading instrument indicating both positive and negative peaks which is absolutely necessary in modern recording. The level calibration is adjustable from the back of the console with a multiturn preset potentiometer. The ledbar reads all signals appearing before the insertion point of the monitor/effect section. Therefore all signals that are audible via monitor p.f.l. This could be the multitrack input, the multitrack sync output, the multitrack remix, or the effect input. It is also possible to use the monitor section as an effect return without having to decouple the ledbar from the multitrack replay (remix) output. Do not patch into the effect input but into the monitor insertion input with effect returns. The ledbar continues to register the multitrack recorder.

Mic

Below the ledbar section aie the input circuit controls and switches. The first being the, per channel, switchable +48V phantom power supply. Below this is the – 20 dB pad, necessary for extremely high input signals on the mic input.



Line

The line switch changes the XLR input to line level sensitivity and also changes the balanced mic input connection into an unbalanced line input. The input sensitivity ranges from – 10 dBu to + 20 dBu.

Remix

The remix switch, which also activates a line level input has priority over the line switch. The remix input is combined with the sync input on the back of the console.

Gain

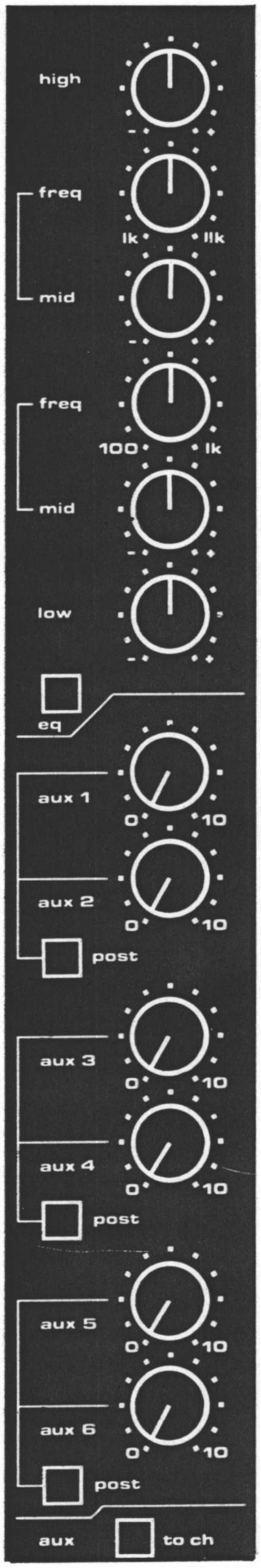
The gain control acts for the mic amplifier as a feedback control and in the line/remix mode as an input attenuator. The mic gain ranges from – 20 dB to – 64 dB whilst providing an enormous headroom with a minimum of 40 dB.

Phase

This phase reversal switch is active on both mic and line/remix inputs which proves handy in all sorts of recording situations.

High Pass Filter

The high-pass filter is a fixed frequency filter with a – 3 dB turn over frequency at 100 Hz. The slope is 9 dB per octave.



Equalizers

The equalizer stands out by virtue of its simple yet effective design, with a minimum audiopath which guarantees a good signal to noise ratio. It is of a parametric 4 band design which spans the whole audio spectrum. The high shelves at 12 KHz and the low at 60 Hz. The high midranges from 1 KHz to 11 KHz and the low midranges from 100 Hz to 1 KHz. The lift and cut range of all 4 equalizer sections is ± 16 dB. The point of turnover frequencies in this equalizer will pleasantly surprise you. In the eventuality of still further equalisation being necessary there follows an insertion point which makes insertion of additional e.q. units possible. The whole e.q. section is bypassable with a silent switch.

Aux

The 4000 series offers in total 6 individual aux sends which easily allows for the most extensive remix sessions. The aux sends are per pair switchable pre/post the monitor/channel fader. Basically the 6 sends are wired pre/post the monitor fader which makes it possible to have foldback as well as effect pre and post the multitrack machine.

Aux to Channel

This switch connects all the 6 sends pre/post the channel fader this being necessary in the remix mode.

Subgrouping

Subgrouping in the 4000 console is done in a new way and demands a new way of thinking from the engineer. The basic idea is to have subgroup amplifiers only where you need them. This means that there is no group amplifier preceeding every multitrack channel as you might have been used to in conventional in-line designs.

In the 4000 console there are only 4 subgroup amplifiers, only 4? Yes, only 4, but these 4 subgroups are fully floating. You can switch them to the inputs as well as to the outputs anywhere in the console.

Imagine, routing from channel 1 to channel 28 without patching, this is possible in the 4000 series in the following way.

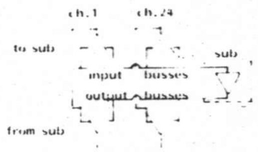
There are 4 switches for this novel routing system, 2 for going to the subgroups and 2 for coming from the subgroups, called "to sub" and "from sub".

If you are not subgrouping, the signal coming from the channel fader goes directly to the multitrack machine.

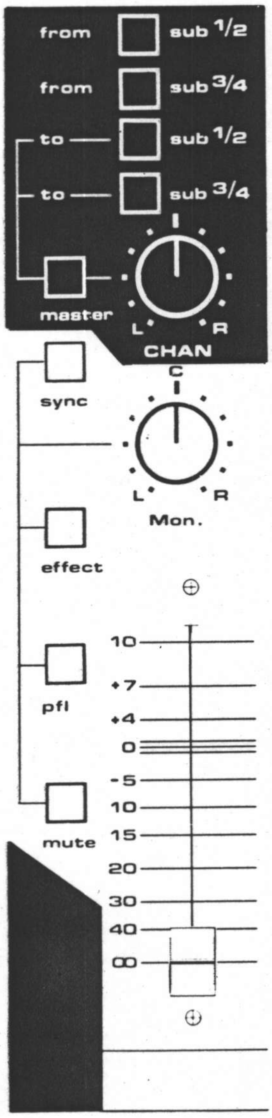
But let's say you want to stereo subgroup channel 1 to 8 to multitrack channel 1 and 2. This means that you need 2 subgroups because you want to do it in stereo. The first thing you have to do is to bring the signals to the subgroups by pushing the switch marked "to sub ½" (you are using now subgroup 1 and 2). The pan-pot determines the signal level sent to subgroup 1 and/or 2, depending on its position, left, right or central.

The signals coming from channel 1 to 8 are now brought to subgroup 1 and 2 (physically located in the master section). But, now, you need them on multitrack channel 1 and 2. You only have to connect the

multitrack inputs to the outputs of the subgroup amps. This is done by switching the "from sub ½" switches. As you can see it is very easy to bring this signal to any multitrack input or even more than 1 multitrack channel, if you like.
For another example you might want to bring channel, let's say 9 – 10 to track 4. Push in, in channel 9 – 10, the "to sub ½"switch (or "to sub ¾"switch, which ever one is available), pan to the right and push the switch in channel 4 called "from sub ½" (or again "from sub ¾" if chosen). You see this is a very flexible way of routing a signal through the console. In fact it is possible to route, without patching, from any input to any output.



This system allows a great saving in switches electronics and labour when compared with the conventional 24 track routing system in a standard in-line console.
Besides sending the signal to the subgroups it is also possible to feed the master section simultaneously.



Pan-Pot

As already described above, this control (with a – 3 dB attenuation when set central) pans the signal between the odd and even subgroups as well as left/right master buss, if selected.

Monitor Section

The monitor section (light coloured) takes care of all the monitoring in the channel.

Sync

The sync switch in the monitor section handles the sync switching of the multitrack recorder. In the on position, the monitor section in the channel is switched from the input to the ouput of the multitrack machine.

Effect

The switch "effect" makes it possible to use the monitor sections as effect returns. In this manner you have control over as many effect returns as the console has channels. The effect inputs are on the back of the console.

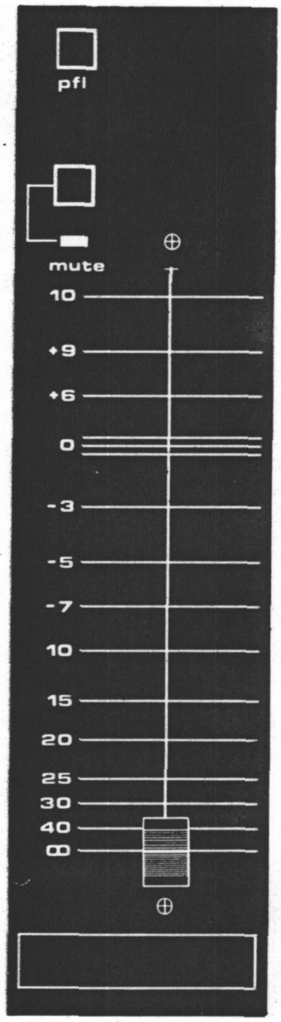
P.F.L. and Mute (Monitor)

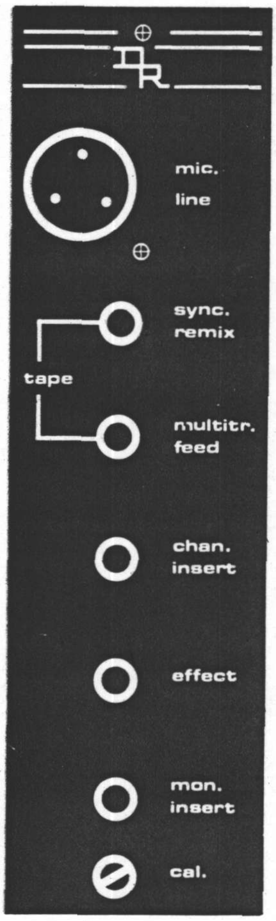
The p.f.l. switch enables you to prefade listen to the signal coming from the channel or from the multitrack as well as from the effect input. The p.f.l. is of the autotype, it switches automatically the stereo master from the monitoring and substitutes it for the activated channel.

Muting is done by cancelling the signal coming from the channel, sync or effect inputs. The p.f.l. is not affected by muting.

Fader Monitor

This small fader, 58 mm travel, is of the carbontrack type. There is a 10 dB gain factor in the amplifier which follows the fader.





P.F.L. and Mute (Channel)

The p.f.l. and mute switches in the channels have the same functions as in the monitor sections. The p.f.l. does not interrupt the signal path. The mute function has a led to indicate its function.

Fader (Channel)

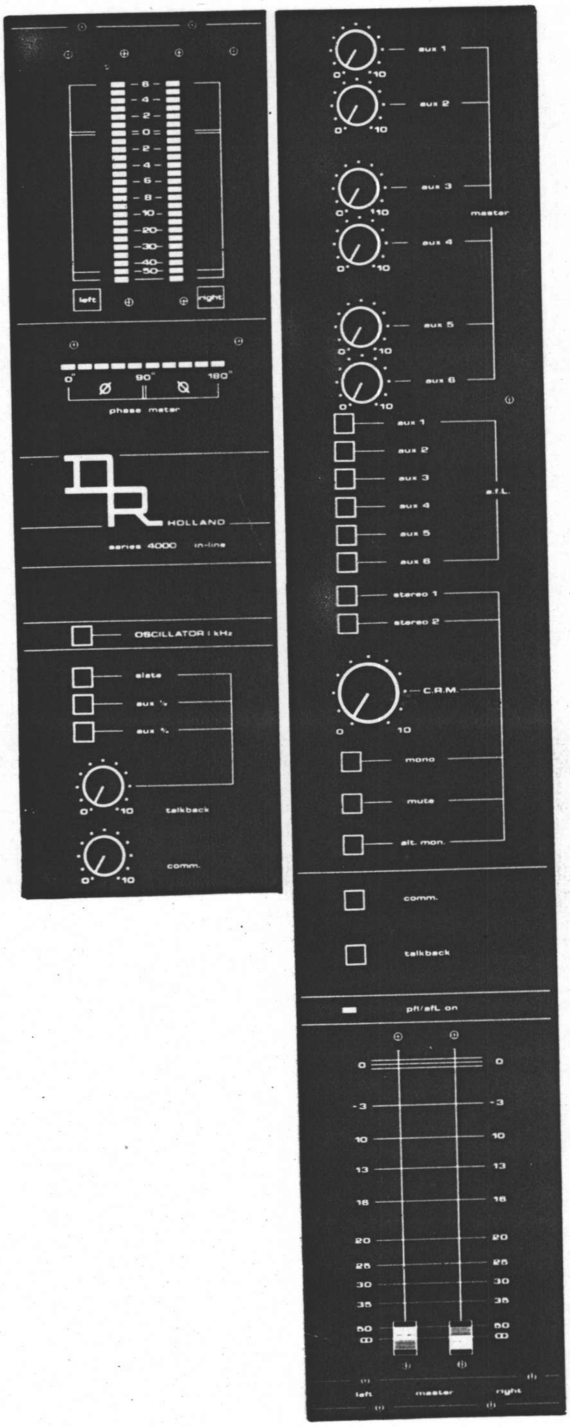
The channel fader is of the carbontrack type with a 100 mm length. Standard is the Noble fader. Options are A.L.P.S. and Penny and Giles. There is also a 10 dB gain factor in the amplifier which follows the channel fader.

Mechanical Strength

The mechanical strength of the individual modules is achieved by using a specially manufactured "U" type profile to which the printed circuit board is firmly secured.

Inputs/Outputs Channel

On the back of the console you will find the in/outputs of the channels. On top there is the XLR type input connector for the mic and line amp. The mic amp is balanced and the line unbalanced. Next there is the combined sync/remix input which accepts two levels of +4 dBu (the professional standard) and the −10 dBV which is the semi-professional standard now-a-days. Multitrack Feed is the output of the channel which has to be plugged into the input of the multitrack machine. On this socket also there are +4 dBu and −10 dBV levels available. We have used the following wiring configuration. On tip of the jack the −10 dBV signal is available and on the ring simultaneously the +4 dBu standard.
By connecting to the appropriate soldertag on the stereo jack plug you choose the tape level you are going to use. The channel insert is the jack into which you can insert ancillary equipment such as compressors noise gates and other frequently used effect devices. The tip is the send and the ring is the return.
The effect jack is there as an extra line level input in the remix situation. The monitor insert is there to make possible extra equalizing or the insertion of other ancillary equipment in the monitoring section of the channel.
The calibration control is there to calibrate the level of the peak reading ledbar. It is factory adjusted to indicate 6 dB down from the signal which is actually measured. This way of calibrating a peak reading ledbar is common practice now-a-days.



Master Section

The following describes from above to below the use and function of the master controls. First there are the extremely precise 25 segment peak reading ledbars with both a positive and negative reading display. The attack and decay characteristics conform to standards used throughout the world. Below these ledbars is the phase correlation meter which is wired in parallel with the ledbar meters. This precision instrument indicates the exact phase relationship between two given signals. A phase shift of 90° degrees or less is acceptable for mono compatability but above this (out of green into red) is unacceptable. The phase meter registers a correct reading between −40 dBu and +20 dBu. When presented with only one signal or with a signal below −40 dBu it switches itself off, so avoiding any incorrect reading.
The oscillator is of the phase shift type which produces a low distortion 1 KHz sinewave.
When its switch is activated a 1 KHz tone is available at all the multitrack outputs at +4 dBu/−10 dBV level, which makes lining up machines a simple procedure.
Then there is the communication section. We at D & R find good communication between the studio and control room an absolute necessity for the success of a recording session. Therefore the 4000 series offers the possibility of comprehensive communication from the studio to the control room at all times and in all stages of a session. It is also possible to speak from the control room to the studio via all outgoing lines by means of the aux outputs.
The slate switch makes it possible to put information on tape.
By use of the talkback switch the C.R.M. is attenuated by 20 dB. There is a high quality, built in, electret microphone for talkback purposes. A high pass filter further increases the clarity.

Aux Masters

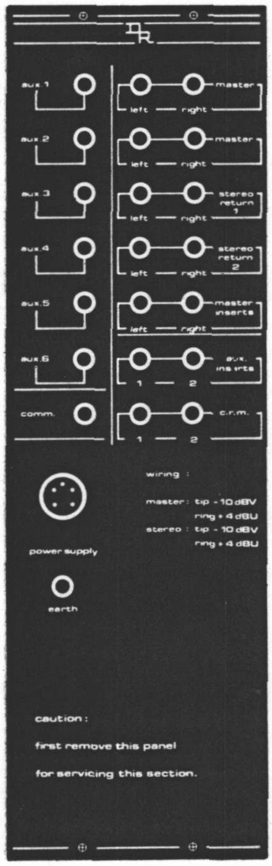
The 6 aux masters with their a.f.l. switches control the total outgoing level of the aux sends.

C.R.M.

C.R.M. stands for Control Room Monitor and regulates the level of all the signals going to the control monitor. A number of push-button switches is found in the neighbourhood of the C.R.M. switch:
The stereo 1 and 2 switch: this switch makes it possible to select from 2 stereo sources instead of the master mix down.
The mono switch makes comparison between stereo and mono possible. Mute cancels the monitor completely. Alt. mon. stands for alternative monitoring. It is possible with this switch to bring in another monitor system if one is connected.
The p.f.l./a.f.l. led indicates whether p.f.l./a.f.l. switches anywhere in the console have been activated.
Noble faders are standard but A.L.P.S. or Penny and Giles are available optionally.

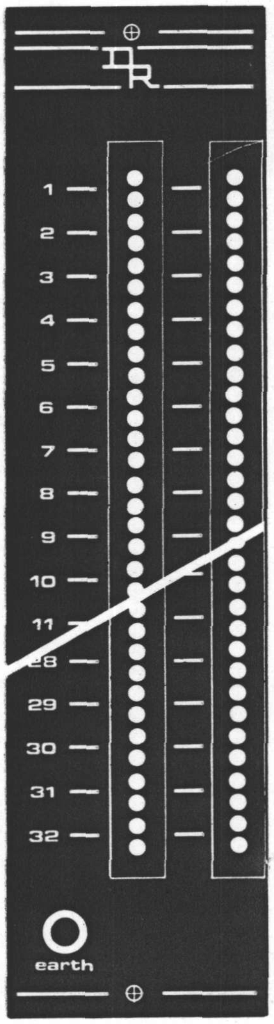
Master Section In/Outputs

There are 21 in/outputs on the master output section. There follows a description of these from top to bottom, beginning with the left hand row as viewed from the rear of the desk. Firstly there are the Aux outputs 1 – 6. All the aux outputs are unbalanced at + 4 dBu. Then below the six aux jack sockets there is a seventh, which is a balanced mic input for communication purposes.
The middle row of master outputs and the row on the right when viewed from the rear are as follows from top to bottom.
Firstly the master left/right outputs with an output level of – 10 dBV on the tip of the stereo jack and + 4 dBu on the ring. Next we have another master output similar to that just described. This second output is intended for use with a second master machine. Below these master-outputs are the 2 stereo returns for the master machine. These stereo jacks have an input sensitivity of – 10 dBV on the tips and + 4 dBu on the rings.
Below the stereo returns are the master inserts and two aux inserts and finally the 2 C.R.M. jacks which deliver the C.R.M. outputs.
Power supply connection is by means of a 5 way XLR connector. The two earth connectors (one below the power supply input socket and one in the patch bay) are to be used when a patchbay is installed in the mainframe to prevent earth loops when patching. Link the two points with a heavy earthing conductor.



Patchbay

The patchbay is modular and has provision for 64 break patch-points per module. The connections between the patch-points and the socket at the rear of the console are made via two printed circuit boards which have at the back molex pin connectors.
Patching is done as follows: It is necessary to solder a stereo jack plug to one end of a twin screened lead (in which each of the conductors is individually screened). To the other end solder a two pin molex female connector. You now have the standard 4000 series internal patchcord.
Now let's say, for instance, you want to wire the channel 1 insertion point to the patchbay. To do this take a 4000 series patch cord (Jack-Molex) plug the stereo jack into the insertion socket on channel 1 and connect the Molex female connector to the pins marked 1 on the Molex strip connector on the rear of the desk. In this way you can bring any in/output jack to the patchbay by means of simple external wiring and further it is an economical method which saves time.



A. Single source on single track-Recording

The microphone or line signal enters the in/output channel, at the point where the mic/line push button will determine the input mode. A gain control for the microphone and line signals is provided with additional -20 dB pad for the microphone signal. Phase reverse can be used on both mic and line signals. If necessary, the 48 volt phantom powering can be switched on.
Firstly the signal goes through the high-pass filter which can be switched in or out. After passing through the high-pass filter it enters the equalizer section which can also be bypassed. After the e.q. the signal appears at the ancillary equipment jacksocket. This provides a send and return path for the introduction of effects or other treatment of the signal. The p.f.l. switch monitors after the insertion point, but before the mute switch. Directly after the mute switch (above the mute led) is the long travel fader. This fader sends the signal via the channel pan-pot to the master/group busses if activated.
In case of a single track recording it is better to by-pass the pan-pot and the group-amps. This is done simply by not activating any of the "to" and "from" switches. In this way only the post fader channel amp is directly connected to the multitrack input. There are two ways of monitoring the signal, directly from the channel:

- by pushing the master routing switch, which leaves the monitor section free for other purposes.
- the 2nd way is to use the monitor fader and its associated pan-pot which feeds the signal directly to the master mix busses, if the monitor mute is not activated.

It is an absolute necessity to mute every unused monitor channel in order to achieve best overall signal to noise ratio in the master mix busses. This muting removes the mix resistors from the buss and thereby provides a lower noise gain. In an unused channel the master switch has to be inactive to achieve best signal to noise performance.
When no effect switch is activated the led bargraph will read the signal going to the multitrack. Aux sends can best be used pre/post from the monitor fader. In this configuration the aux sends are also available in the sync mode. If you wish to monitor from the channel you have to push the aux to channel switch.
The standard way to record in one channel is to drive the multitrack machine as hard as possible and to monitor the signal through the small fader. Both in the recording mode and in the sync mode.

B. Multiple sources on one or two tracks

When more than one microphone or line signal has to be recorded on a single track or on two tracks for stereo, a submix facility will be required.
On the 4000 series this can be done, simply, without patching. The microphone or line signals will be processed as described under A, except that one of the two sub switches must be activated. If you push, for

example, routing switch "to sub ½" in channel 1, the signal will, depending upon the position of the channel pan-pot, go to subgroup 1 and/or 2.
To bring the output of this subgrouped channel to multitrack channel 1, it is only necessary to activate the switch "from sub ½" in channel 1. Now the subgrouped signal replaces the channel signal. In this way you can route as many channels to the subgroup amps as you wish and connect the output to any multitrack input in the console wherever you like. It is possible to make 2 stereo subgroups at the same time and to bring these to 4 multitrack channels. If more subgroups are required you can use the 2 stereo subgroups as 4 mono subgroups by using the pan-pots.
When taking signal from the subgroups, note that odd numbered channels are fed by subgroup outputs 1 and 3 and even numbered channels by subgroup outputs 2 and 4.

Sub

Imagine you have a mix of 12 channels in perfect balance routed to track 1 + 2 and because the overall signal level is to high for the tape machine it becomes necessary to attenuate the mix. This can be done as follows.
De-activate the "from sub ½" switches in channel 1 + 2 and activate the same numbered switches in two other unused channels.
From these unused channels you can now patch from the monitor insert sends into another two unused channels. In these two channels activate "to sub ¾" and activate the "from sub ¾" in channels 1 + 2.

Sync

The sync replay is done simply by activating the sync switch. The monitor faders handle the sync replay signals.

C. Overdub

When a small part of an allready recorded track has to be re-recorded, several complications arise due to the fact that for monitoring and cueing purposes the track has to replayed in the sync mode before and after the part to be re-recorded, whilst during the re-recording the channel should function as for normal recording (as described under A). When the recorder gives the input signal on its outputs (if not in the sync or replay mode), the following simple set-up is preferred.
Process the microphone or line signal as under A, but with the channel in the sync mode. Now the engineer can listen to the sync signal coming from the machine which tells him at which moment he will go into overdub and as soon as the multitrack goes into the record mode he hears the musician playing or singing. If the engineer would like to hear the musician also before and after the dubbing he only has to push the master routing switch.
On the other hand the musician himself has to hear the

sync signal as a guide as to where to start dubbing and he wants to hear himself before dubbing.
The set-up for this situation is as follows. Use another input channel and give the studio foldback from for instance aux 1. The musician now hears himself (activate "aux to channel" first) via this channel's foldback facilities.
Route this signal to the multitrack and place the multitracks sync signal also on the aux 1 buss. The musician will hear himself continuously before, during and after going into the recording mode.

D. Remix

When all tracks have been recorded to full satisfaction, the final end mix will have to be made. All the remix switches have to be activated and the master switch. "Aux to channel" has to be activated too. This is a basic set up for 8 – 16 – 24 tracks into 2. Ensure that unused channels have their master switches in the off position and that the muting switches in the monitoring section are in the "mute" mode to optimise signal to noise ratio.
Each channel monitor section (light coloured) has an effect return. This can be used as an extra input in the remix. These can be used in combination with the mic/line inputs. A 24 channel desk then provides 48 inputs.
Subgrouping in the remix is easily done by removing those signals from the master busses (de-activate the "master" switch) which have to be subgrouped and routing them to the subgroups.
In those channels where you activate the "from sub" switches you have a pre-master subfader in the monitoring section.
By following the signal path in the block-diagram the aforementioned situations will become clear.

Servicing

Servicing a module is very easy. Firstly remove all the in/output jacks from the back of the channel. Then unscrew the back panel and take it out of the console after having disconnected the XLR plug. Now remove the 24 conductor molex flatcable and the 6 conductor flatcable. Next step is to unscrew the module and take it out of the mainframe. In the master section you have to unplug the power XLR plug before taking out the master module.

Summary

It should now be apparent that the D & R 4000 series in-line consoles represent a novel approach to console design. We should also like to bring to your attention the following facts.

Console construction is made from steel which gives an excellent shielding from crosstalk and external R.F.I. or E.M.I. (noise).
As already stated this 4000 series is fully modular. This means that any mainframe can accept 10 modules of 47 mm width. The master module is twice this width. As an example 1 mainframe can accept:
- 8 in/output modules and one master
- 4 in/output modules 4 blank panels and one master
- 6 in/output modules 1 master and 2 patchpanel modules.

2 mainframes can accept:
- 16 in/output models 1 master module 2 patchpanel modules.

3 mainframes can accept:
- 24 in/output modules 1 master module 4 patchpanel modules.

All modules are fully interchangeable in the mainframe to realize any combination you have in mind.

Options

Optional extra's are available if desired
- e.q. on/off leds
- p.f.l. on/off leds
- from sub on/off leds
- monitor mute on/off leds
- A.L.P.S. professional carbon track faders
- Penny & Giles conductive plastic faders
- Stands
- Fader and Mute automation system "SCORE" (Studio COmputer REmix).

Ordering Information

Quant.	description	costs	total
	Mainframe		
	subframe		
	channels		
	master		
	blindpanel		
	patchpanel		
	patch cords int.		
	patch cords ext.		
	faders opt.		
	leds e.q.		
	leds p.f.l.		
	leds from sub.		
	leds mute		
	stands		

system costs

SPECIFICATIONS

Notes: Nominal operating level throughout the console is 0 dBu (o. 775 v) — Nominal output level is + 4 dBu/-10 dBu.

Microphone Preamplifier

electronically balanced R.F. suppressed.
input impedance 2 kOhm
gain +64 dB to +0 dB (44 dB variable gain with 20 dB "pad")
headroom min. 40 dB. Max input +20 dB.
noise −126 dB (A weighting)

frequency response referred to 0 dB at 1 kHz
−0.5 dB at 20 Hz 0.5 dB at 40 kHz −3dB at 110 kHz
Harmonic distortion

with −20 dBu input 0 dBu output	with −20 dBu input +20 dBu output
0.0084% at 50 Hz	0.0086% at 50 Hz
0.0081% at 1 kHz	0.006 % at 1 kHz
0.0095% at 10 kHz	0.018 % at 10 kHz

Line/Remix Amplifier

input impedance 10 kOhm
gain from −10 dBu to infinity
headroom 22 dB
Equivalent input noise
−96.5 dB (20—20.000 Hz)
frequency response referred to
0 dB at 1 kHz
−0.5 dB at 8 Hz
−0.5 dB at 140 kHz
−3 dB at 400 kHz

Harmonic distortion

0 dBu output	+20 dBu output
0.0074% at 50 Hz	0.0074% at 50 Hz
0.0024% at 1 kHz	0.0016% at 1 kHz
0.0058% at 10 kHz	0.0048% at 10 kHz

Equalizer Section

±16 dB at 12 kHz
±16 dB from 1 kHz to 11 kHz with Q factor 1.5
±16 dB from 100 Hz to 1 kHz with Q factor 1.5
±16 dB at 60 Hz
high pass 100 Hz
slope 9 dB per octave

distortion all filters unity gain	all filters + 16 dB
0.0074% at 50 Hz	0.0017% at 50 Hz
0.0062% at 1 kHz	0.008 % at 1 kHz
0.013 % at 10 kHz	0.0016% at 10 kHz

Overall Performance

sync/effect input impedance 10 kOhm
sync sens. +4 dBu/-10 dBu. Effect sens 0 dBu.
Output impedance 100 ohm on all outputs
max output +22 dB into 1 kOhm and above

Record Mode

Test condition; One channel, assigned to and from groupbuss output, microphone input loaded with a 150 ohm source, mic preamp set for 30 dB gain group output +4 dBu
frequency response referred to 0 dB at 1kHz −0.5 dB at 20 Hz and 20 kHz

distortion
0.017 % at 50 Hz
0.0094% at 1 kHz
0.052 % at 10 kHz

noise
−84 dBu below +4 dBu output (20—20.000 Hz)

Mix Mode

frequency response
−0.5 dB at 17 Hz
from line inputs to stereo mix buss outputs
ref to 0 dB at 1 kHz
−0.5 dB at 40 kHz
−3 dB at 135 kHz
distortion no more than 0.009% at 1kHz
headroom +22 dB, output amp +18 dB

noise −84 dB below +4 dBu (20—20.000 Hz) measured at the stereo buss outputs with stereo master fader at max. all channel faders at full attenuation panpots at their center positions
−83 dB below +4 dBu (20—20.000 Hz) with one channel fader at unity gain

Crosstalk

Record mode
Direct Assign between two channels both at 30 dB gain +4 dBu out of channel 1, 150 ohm source on channel 2 input.

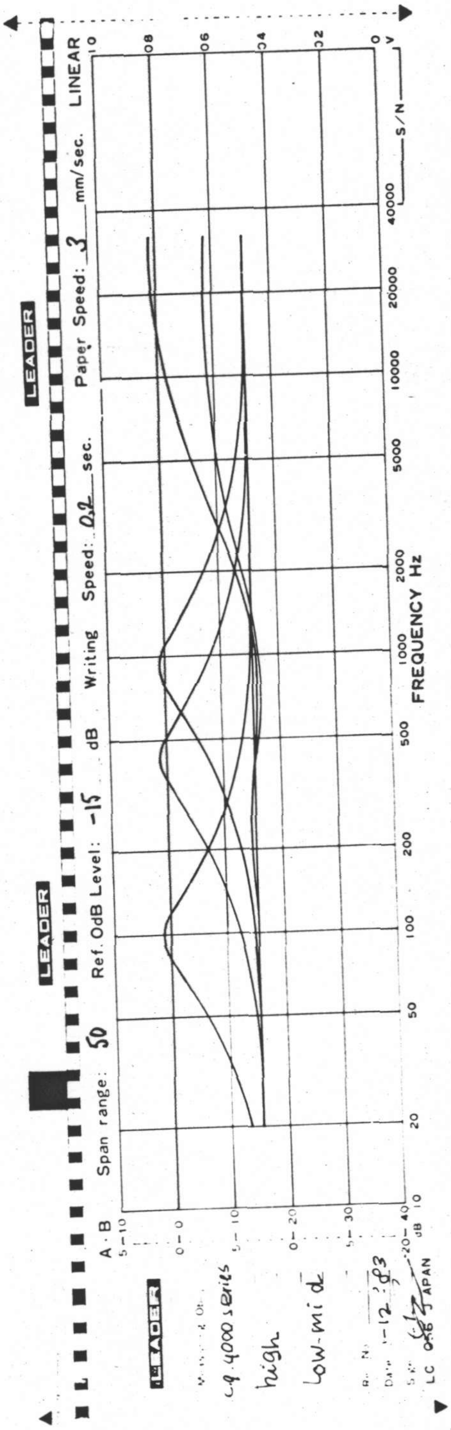
Crosstalk on channel 2 (referred to +4 dBu)
100 Hz better than −88 dB
1 kHz better than −90 dB
10 kHz better than −74 dB

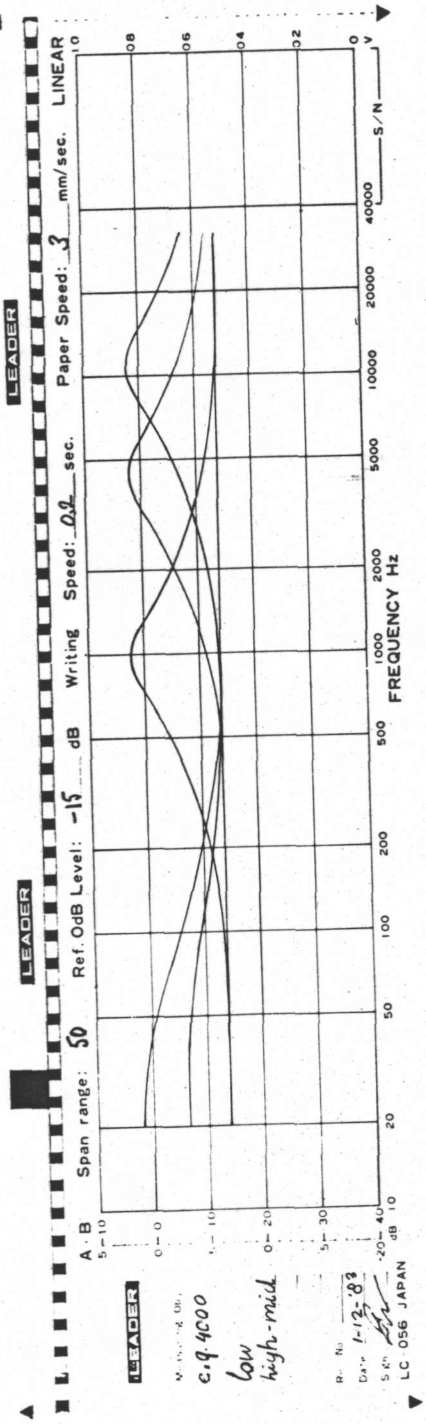
Mix Mode

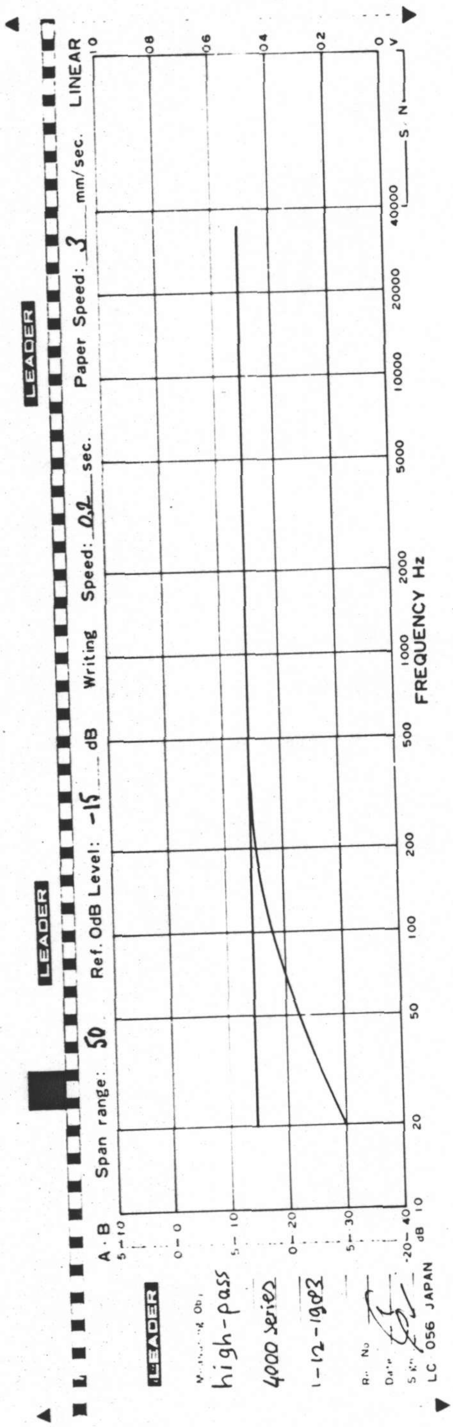
Channel 1 is fed with +4 dBu, fader at unity panned to left. Stereo master fader at maximum.
Channel 2 is terminated with a 20 ohm source.
Fader at unity, panned to right stereo master.

Crosstalk on right master output.
100 Hz better than −77 dB
1 kHz better than −70 dB
10 kHz better than −63 dB

Typical equalizer and filter curves







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 cableto 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 carefully read the followig information

Especially in sound equipment on stage the fllowing 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 is primarily determined by the amount of current flowing through a person.h**

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. Atypical 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 pofusely 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 guitar/synth 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 only be cured by proper 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.

Always keep the above mentioned information in mind when using electrically powered equipment.

HANDLEIDING

SERIES 4000

nederlands

PAN POT-
Deze knop bepaald per kanaal de plaats van het signaal in het stereo beeld. Dat is van volledig links/rechts tot elke positie ertussen. Het signaal wordt dus verdeeld tussen de even en oneven subgroepen alsmede de links/rechts masteruitgangen. Wanneer de pan pot centraal staat wordt het signaal 4,5 dB verzwakt.

MONITOR SECTIE-
De monitor sectie is wit. Het vergemakkelijkt de bediening en maakt het paneel overzichtelijker. De volledige monitoring in een kanaal wordt door deze sectie geregeld.

SYNC-
De sync-schakelaar in de monitorsectie bepaald het opnemen cq weergeven van de meersporen recorder in de monitorsectie. Wanneer de schakelaar ingedrukt is geeft dit aan, dat de monitor sectie in het kanaal automatisch wordt geschakeld van de ingang naar de uitgang van de recorder.

EFFECT-
De effect-schakelaar stelt U in staat de monitorsectie als effect return te gebruiken. Hierdoor heeft U net zoveel effect returns als de tafel kanalen heeft.
De effect ingangen vindt U op de achterzijde van de mengtafel.

PFL en MUTE (monitor)
Via de pfl schakelaar luistert U vóór de fader naar het signaal dat komt vanaf de kanaalfader of van de meersporen recorder, alsmede de effect ingang. Deze pfl is auto-type, dat betekent dat de pfl automatisch de stereomaster van de monitor afschakelt en deze vervangt door het gekozen kanaal.
De mute toets onderbreekt het signaal, komende van de multitrack sync of effect inputs. De pfl wordt niet door de mute beinvloed.

FADER MONITOR-
In de monitor sectie is een kleine, 58 mm, carbontrack fader gemonteerd. U heeft 10 dB versterking ter beschikking in de versterker die na de monitorfader volgt.

EQUALIZERS-

De equalizer van de series 4000 is niet gecompliceerd maar wel uitermate effectief. De signaalweg is minimaal waardoor een uistekende signaal ruis verhouding verkregen wordt. De viervoudige equalizer omvat het hele audiospectrum. De laag cq hoogsectie heeft zijn kantelpunten op resp. 60 Hz en 12 kHz. De high-mid toonregeling loopt van 1 tot 10 kHz terwijl de low-mid een bereik heeft van 100 Hz tot 1 kHz. De kantelpunten zijn zeer muzikaal ontworpen; ze zullen U verrassen!
Mocht extra toonregeling nodig zijn: een insertie punt is gemonteerd direct na de toonregelsectie. De gehele toonregeling op een kanaal kan d.m.v. een bypass-schakelaar overbrugd worden. Een led geeft aan of zij al dan niet is ingeschakeld.

AUX-

De series 4000 heeft in totaal 6 effectsends te bieden welke ruim genoeg zijn voor de meest uitgebreide remixsessies. Ze zijn per paar pre/post de monitor fader schakelbaar. Wanner de aux-to-channel wordt ingedrukt zijn de effectsends pre-post de kanaalfader schakelbaar. Staat deze schakelaar niet ingedrukt dan is het mogelijk pre/post de monitorfader een signaal aan de muzikanten in de studio aan te bieden. (=foldback) Mede is het mogelijk een effectapparaat aan te sturen voor of na het signaal van de multitrack recorder.

AUX-TO-CHANNEL-

Deze schakeling zorgt ervoor dat alle aux sends pre/post de kanaalfader schakelbaar zijn, iets wat tijdens de remix-sessie noodzakelijk is.

SUBGROUPING-

Het maken van subgroepen in de series 4000 gebeurt op een vrij onorthodoxe manier en verlangt een andere manier van denken van de technicus. Het basisidee is om alléén daar subgroepen te hebben waar men ze ook nodig heeft. Dit betekent dat het mogelijk is op ieder gewenst kanaal een subgroep te vormen. D&R noemt dit FSS; Floating Subgroup System! Se series 4000 heeft 4 subgroep versterkers. Dat is voldoende omdat de subgroepen dus overal inzetbaar zijn. U kunt de subgroepen naar de ingangen maar ook naar de uitgangen overal in het mengpaneel schakelen. De routing van het kanaal wordt verder beschreven onder het hoofdstuk ' Meerdere Signalen op een of twee Sporen'. Dit zwevende subgroep systeem bespaart erg veel schakelaars en andere componenten vergeleken met andere in-line mengtafels. De bedrijfszekerheid wordt hierdoor allen maar vergroot. Uiteraard is het mogelijk om naast de subgroepen de mastersectie gelijktijdig aan te sturen.

BESCHRIJVING VAN DE BEDIENINGSORGANEN:

LED BAR-

De 11 segments led bar (de onderste led geeft aan of de tafel is ingeschakeld) is een peak led bar die zowel positieve als negatieve peaken aangeeft; iets wat absoluut noodzakelijk is in moderne studiotechnieken. De level calibratie is instelbaar aan de achterzijde van het kanaal. De led bar geeft de niveaus van alle signalen weer welke vóór het insertiepunt van de monitorsectie de tafel binnenkomen. Dat zijn dus alle signalen die U kunt beluisteren via de monitor pfl. Dit kunnen zijn: de multitrack input, de multitrack-sync output, de multitrack remix of de effect-input. Het is ook mogelijk de monitor sectie als effectinput te gebruiken zonder dat de led bar wordt losgekoppeld van het recordersignaal. Patch in dat geval niet in de effectinput maar in de monitor insertie jack. Het blijft dan mogelijk het niveau van het recordersignaal af te lezen.

MIC-

Onder de led bar vindt U de bedieningsorganen van de microfoon ingang. De bovenste schakelaar geeft de mogelijkheid de 48 V phantoomvoeding per kanaal aan/uit te schakelen. Hieronder bevindt zich de -20 dB schakeling welke gebruikt kan worden om oversturing van het microfoonkanaal te voorkomen.

LINE-

De line schakelaar schakelt de microfooningang om naar line-gevoeligheid en impedantie. (resp. -10 dBu/+20 dBu, 10 kOhm)

REMIX-

In feite is de remix-schakelaar niets anders dan een line-schakelaar met een prioriteit boven de normale line ingang. De remix input is gecombineerd met de sync-input op de achterzijde van het paneel.

GAIN-

De gain regelaar regelt de mate van versterking of verzwakking van het aangeboden signaal. De microfoongevoeligheid reikt van -20 dB tot -64 dB, met een uitsturingsmarge van meer dan 40 dB.

PHASE-

De phase omkeerschakelaar werkt zowel op de microfoon- als de op de line-input.

HIGH PASS FILTER-

Het laag af filter heeft een vaste frequentie die ligt op 100 Hz. De afval van het filter is 9 dB per octaaf. Ongewenst laag kan op deze manier effectief worden weggefilterd.

Wij danken U hartelijk voor Uw keuze en het vertrouwen dat U in ons produkt stelt. De D&R series 4000 mengtafel is ontworpen door en voor professionele gebruikers. Het geeft U een uniek in-line consept in handen met een enorme flexibiliteit. Onze reeds twaalf jaar lange ervaring in het ontwikkelen en produceren van geluidsmengtafels resulteerde in een betrouwbare en bedrijfszekere mengtafel. Lees de gebruiksaanwijzing eerst aandachtig door om het volle profijt uit Uw mengtafel te halen. Schroom niet om de aangewezen mogelijkheden te gebruiken; de megtafel is er voor ontworpen!

Een beknopte lijst van de 4000 features:

- 11 segment led-bar per kanaal
- 48 V phantoom, per kanaal aan/uit schakelbaar
- phase reverse voor zowel mic- als line ingang per kanaal
- extreem ruisarme mikrofoonvoorversterkers
- gecombineerde sync- remix inputs op +4 dBu/-10 dBV
- 100 Hz high pass filter
- 4 bands sweep equalizer
- 6 aux sends pre/post fader schakelbaar
- floating subgroups
- sync- en effectinputs per kanaal waardoor een dubbel aantal kanalen gemixed kan worden.
- monitor mute en p.f.l.
- kanaal mute en p.f.l.
- 100 mm kanaal fader, 60 mm monitor fader
- 2 inserts per kanaal
- multitrack outputs op +4 dBu of -10 dBV
- master sectie met 25 segments led-bar en phasemeter
- 1 kHz line up toongenerator
- talkback met ingebouwde electret-mikrofoon
- communicatie systeem
- 6 master aux sends met afl afluistering
- uitgebreide monitor sectie: twee paar monitors schakelbaar, mono en mute schakelaars.
- twee stereo master recorders kunnen worden afgeluisterd (+4/-10 mogelijk)- 64 jacks modulair patchpanel
- alle connectors via xlr of stereo steek (6,3 mm)