

THE D&R 8000 SERIES 2 IN-LINE SYSTEM

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DESIGN HIGHLIGHTS INTRODUCING

- new mic pre-amps
- new front panel lay-out
- new aluminium "U" profile making for an extremely stable channel construction
- fantastic redesigned optics, especially the recessed ledbars behind the smoke coloured perspex.
- new p.c.b. lay-out for lower crosstalk
- ready for automation

A completely new approach to limiting of above audio range frequencies, through passive filtering (instead of the standard active filtering) gives this console an incredible transparency through its absence of transient distortion.

By critically damping every integrated circuit at 40 kHz square-waves we achieved complete elimination of overshoot and/or ringing and slewing.

As has been mentioned, the subaudio frequency drop is only achieved by passive filtering. This way of designing the electronics in a recording console contributes to a superb transparency throughout the audio range.

The electronics are performed by the well known TL 070 series Bi-F.E.T. integrated circuits from Texas Instruments which are low noise and very fast op amps with a slew-rate of 13 *V/us*.

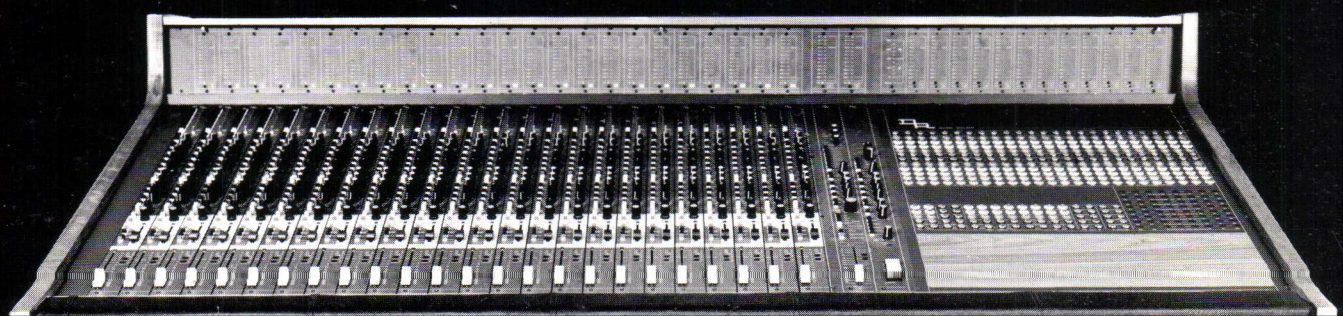
Beside these famous TL series we also use the Signetics NE 5534 AN which is an industrial standard in low noise console design. The passive components are all of a high quality standard. The pushbutton switches have fork contacts and are selfcleaning in operation.

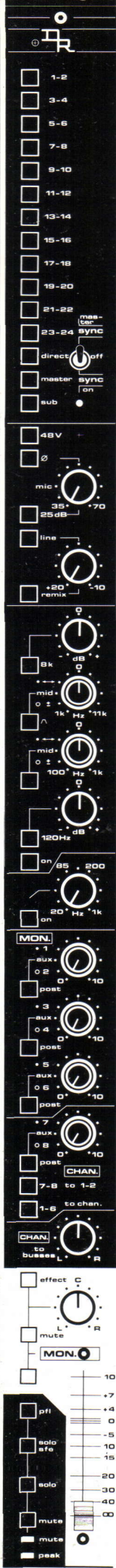
A new earthing system also contributes to a very stable design with extremely good signal to noise ratio, especially in the mixing buss amps.

8000 Series II, a short run down of its possibilities.

- 24 Low noise multitrack summing amps.
- Simultaneous routing to master mix down buss, direct output and multitrack summing amps.
- Centralized sync switching.
- One button subgrouping without patching.
- Phantom powering, switchable per channel.
- New extremely low noise, electronically balanced, mic amps, outperforming every transformer balanced mic amp.
- Click free phase reverse and active 25 dB pad.
- 4 band parametric e.q. of novel design without interacting controls.

- High pass filter from 20 Hz to 1 kHz fully adjustable, and 12 dB per octave.
- 8 aux sends pre/post switchable and selectable from channel and monitoring.
- "One channel dubbing" with full monitoring for engineer and musician.
- A minimum of 24 effect returns incorporated.
- Monitor mute and p.f.l.
- Channel mute and p.f.l.
- Solo in place system with master mode switching.
- 2 Programmed muting systems.
- 100 m.m. carbon track A.L.P.S. faders (conductive plastic optional).
- 12 segment led bargraph peak meter per channel.
- 2 insertion points per in/out channel
- Master section with trimmable left/right outputs, as well as p.f.l./a.f.l. trimming.
- Selection from 6 stereo sources.
- Control Room Monitor.
- A stereo cue system fed by 8 aux sends with complete controlability of stereo balance and level via the control room monitor.
- A high quality function generator with fixed frequencies for tape level adjustments.
- 8 Master aux sends with possibility for audible and visual control.
- Led bargraphs for aux master sends and two cue systems as an optional extra.
- Comprehensive control room monitor section with alternative monitor loudspeaker switching, dim switch, mono switch and separate left/right muting.
- Separate studio monitoring which can be fed from the C.R.M. or the cue 1 system.
- Extensive talkback facilities with built in electret mic and individual selection to slate, STM, cue 1, cue 2 and patchbay (where desired patching connections are made).
- Separate communications system switchable via the a.f.l./p.f.l. system.
- Phase meter standard.
- Internal patchbay with all in-outputs plus 24 external equipment in-outputs.
- All connections via multipin plugs, male plugs being delivered in advance of the console.





ROUTING

The 12 upper push button routing switches facilitate signal routing from the channel fader and pan-pot to busses to one or more multitrack summing amplifiers.

The last pushbutton switch facilitates the creation of a subgroup master fader for all signals routed to the channel concerned.

SYNC

In the position "sync-on" the monitor section in the channel is switched from the input to the output of the multitrack machine. In the position "Master sync" this only happens after activating of the sync in the master section. This provision simplifies the simultaneous switching of more than one track.

MIC

Under the routing section are the input circuit controls and switches. The first being the, per channel, switchable 48 volt phantom power supply. Thereunder is the phase reverse switch which only affects the microphone signal. Following this comes the mic gain control which has a range of 35 dB which can be extended by the active 25 dB "pad". Thereby allowing for more subtle adjustment without degrading the signal to noise ratio. Through the high input impedance of the microphone pre-amp (4 kOhm) it is possible, without any problems, to feed in a line signal with the "pad" switched in. You have then the possibility of a symmetrical line input with phase reverse switching.

LINE

Under the mic gain is the combined line/remix attenuator with a range of, from -10 dBu tot +20 dBu.

EQUALIZERS

The equalizer stands out by virtue of its simple design yet innumerable possibilities. It is of a parametric 4 band design which spans the whole audio spectrum. The high shelves at 2 clickfree -3 dB turnover points at 8 kHz and 12 kHz. The maximum lift and cut is ± 16 dB. This being linearly spread over the potentiometer range.

The two equalizers are parametric. The high mid range, from 1 kHz to 11 kHz with a choice of two bandwidths. The low mid control ranges from 75 Hz to 1 kHz and likewise a choice of two bandwidths. The lift and cut range is ± 16 dB. The bass control is again a shelving control as opposed to the bell curves of the midrange equalizers. The turnover points are at 60 Hz and 120 Hz.

HIGH PASS FILTER

The continuously variable high passfilter is switched immediately before the equalizer.

AUX

The 8000 series offers in total 8 individual aux sends which easily allows for the most extensive remix sessions. Nevertheless the electrical switching circuits require some explanation. A priority in this design was

that, as well as the input channels, the monitor section should be able to make use of the aux sends. This is realized as follows: Aux 1 - 6 are connected to the monitor fader each pair being pre-post switched. Aux 7 - 8 are connected to the channel fader and are likewise pre-post switchable, per pair.

7-8 TO 1-2

When dubbing it is necessary to allow the musician to hear himself as well as the sync signal (which comes from aux 7 - 8 fed from the channel fader (the mic input signal) to the mix busses of aux 1 - 2. In this way the musician hears the signal from the monitor section (through Aux 1 - 2) as well as the input signal from the channel section.

1-6 CHANNEL

Through the switch 1 - 6 to channel all aux sends are available to you in the remix mode.

PAN-POTS

The channel to busses pan-pot is the one associated with the channel fader and is operational on multitrack mixbusses 1 - 2 and master left-right. The monitor pan-pot is associated with the left-right master busses but only if the adjacent mute switch is inoperative.

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.

P.F.L. (CHANNEL)

The p.f.l. is wired immediately after the channel insertion point and before the channel fader. The p.f.l. circuit is not interrupted by the "solo in place" system or the mute switch.

"SOLO IN PLACE"

The system is a centrally operated muting system. As opposed to an a.f.l. system (after fader listening). The Solo in Place system leaves the effect returns unaffected, whereby it is a simple matter to make a comparison between the original and the effect signal. Every muting is indicated by a led.

MUTE

The mute switch has priority over the solo in place system even when a solo safe button is activated. The Mute function is accomplished by means of F.E.T. switching.

PEAK INDICATOR

The peak indicator is wired directly after the e.q.-section and responds to positive as well as negative wave forms.

FADER

The standard channel fader is a carbon track A.L.P.S. fader.

LEDBARGRAPH

The 13 segment led bargraph is a peak reading instrument with adjustable release (0

the p.f.l., which can be set to your requirements.

MASTER SECTION

First and foremost are the Master Trim Pots which offer the possibility of correcting an over, as well as under modulated mix.

THE P.F.L./A.F.L. TRIMPOT

This allows for adjustments of the audible level of the signal coming from one of the p.f.l./a.f.l. switches.

MONITOR SOURCE

The monitor select switches are the controls of the high quality function generator.

CUE

The cue 1 system is identical with the cue 2 system. They are intended as foldback for the musicians. The cue systems are offered with the optional extra of 4 led bargraph meters.

AUX MASTERS

The Aux masters with their a.f.l. switches control the total outgoing level of the Aux sends. Also for these, as optional extras 8 led bargraph meters are available.

C.R.M.

The C.R.M. is surrounded by various push-button switches. Firstly "Alt" which stands for alternative. It is possible with this switch to bring in another monitor system in the

switch attenuates t dB. The mono switch comparison possible

S.T.M.

S.T.M. stands for S monitor system for cue systems.

COMMUNICATIO

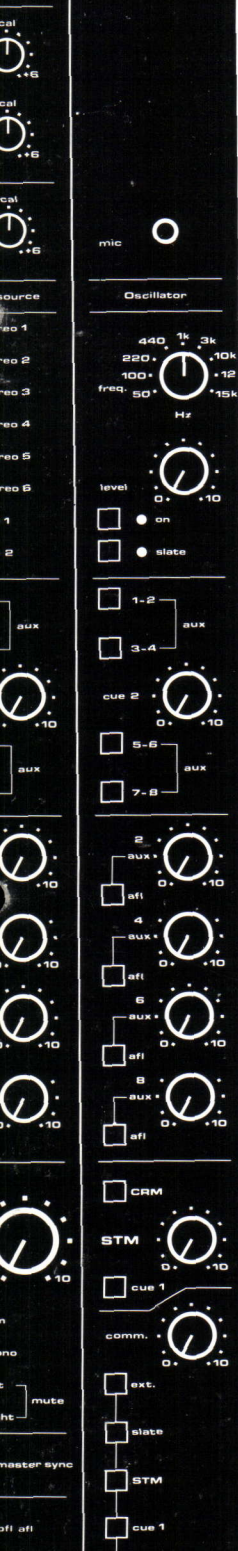
The 8000 series all mic input which via switch and associated be monitored on the

TALKBACK

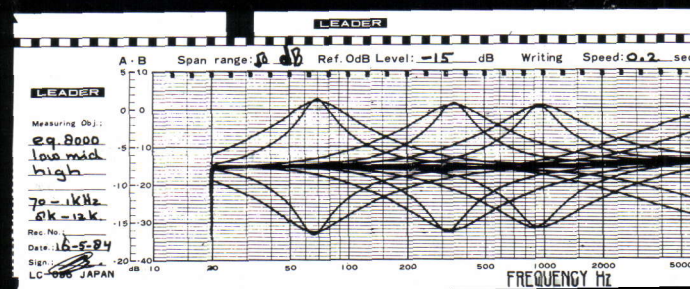
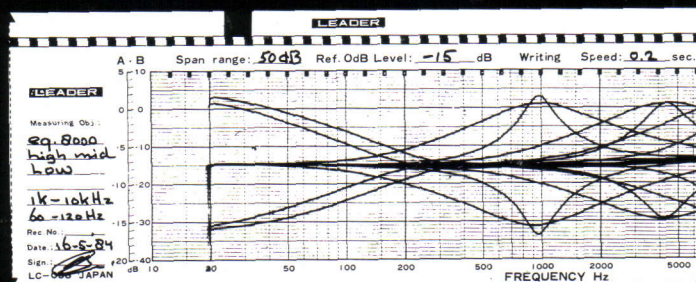
The talkback system we at D&R find good absolute necessity recording session.

PROGRAMMED

This switch (when pushed) switch that cancels channel. It is now mute in the channel now lit up. This operation gives you the solo in place switch programmed mutings in continuously: Consider one instrument on



TYPICAL EQUALIZER AND FILTER CURVES



Notes: Nominal operating level throughout the console is 0 dBu (0.775 v)—Nominal output level is +4 dBu.

MICROPHONE PREAMPLIFIER

electronically balanced R.F. suppressed.
input impedance 4k4 Ohm
gain +70 dB to +10 dB (35 dB variable gain with 25 dB "pad")
headroom +22 dB. Max input +12.5 dB
noise -128 dB (A weighting)

LINE/REMIX AMPLIFIER

input impedance 18 kOhm
gain from -10 dBu to infinity
headroom 22 dB
Equivalent input noise
-96.5 dB (20-20,000Hz)
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

EQUALIZER SECTION

± 16 dB at 8 kHz and 12 kHz
± 16 dB from 1 kHz to 11 kHz with Q factor 1.0 and 2.0
± 16 dB from 75 Hz to 1 kHz with Q factor 1.5 and 2.5
± 16 dB at 60 Hz and 120 Hz
high pass from 20 Hz to 1 kHz
slope 12 dB per octave

OVERALL PERFORMANCE

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

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 1 kHz
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

MIX MODE

Channel 1 is fed with +4 dB, 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 -62 dB

RECORD MODE

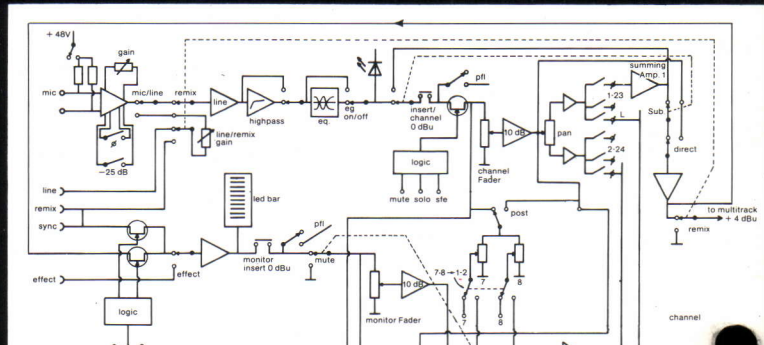
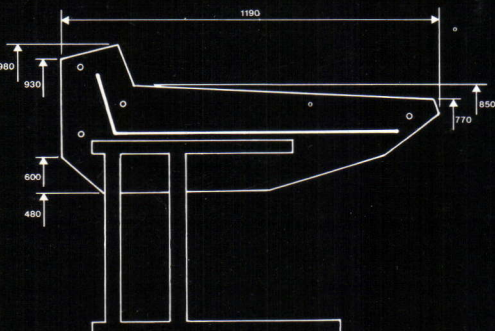
Test condition; One channel, assigned to its same numbered groupbuss output, microphone input loaded with a 150 ohm source, mic preamp set for 30 dB gain, group output +4 dBu frequency response referred to
0 dB at 1 kHz -0.5 dB at 20 Hz and 20 kHz
noise -82 dBu below +4 dBu output (20-20,000 Hz)

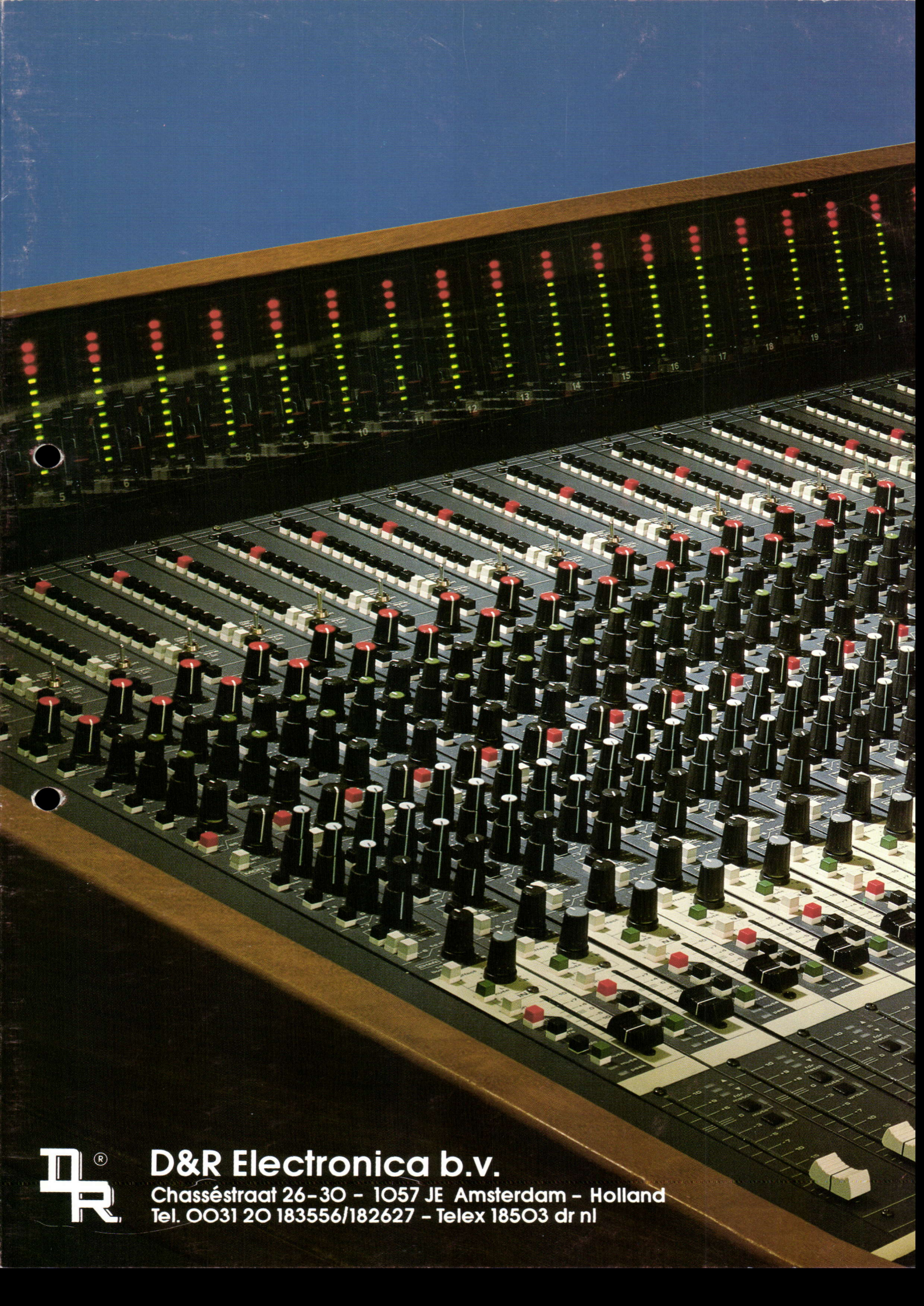
CROSSTALK

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 -79 dB





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