1.1 <u>INTRODUCTION</u>

The D&R DAYNER is an extremely compact and easy to handle mixing console which incorporates all necessary facilities for recording and public address.
It uses our Floating Subgroup System, which makes it possible to create subgroups where necessary.
The DAYNER uses an innovative approach to cope with the conflicting needs which arise when multitracking needs to be combined with public address.

The basic idea behind the DAYNER is to have an input module for every signalsource, be it a microphone signal, line signal, tape output or effect signal, they will all be connected to the DAYNERS mic/line inputs.

To become familiar with all the facilities of the DAYNER we advise you to read this manual very carefully. It will give you important information about operating, installation and service.

D&R Electronica B.V.

1 CHANNEL OUTPUT SECTION

The DAYNER uses our unique Floating Subgroup System, to create subgroups anywhere in the console and only there where it is needed. This is a simple and efficient way of using subgroups.
The upper section of the channel is completely dedicated to subgrouping. When no push-button switch is activated the channel output receives its signal from the post fader-channel amplifier. When one of the a-b-c-d switches is activated, the channel output signal will be replaced by the output of one of the 8 subgroup amps located in the mastersection. Only one at the time is possible. In the uneven channels subgroup 1-3-5-7 only and in the even channels subgroup output 2-4-6 or 8.
The total outgoing level can be adjusted by the trim control (e) over a range of more than 20 dB.
The mix button (f) sends the channel output signal to the master output of the DAYNER. Signals from the <u>odd channels to the left master-output</u> and signals coming from the <u>even channels to the right master-output</u>. This could be the subgroup signal or the channel output. There are two ways of routing signals to the masters. From the output section (sub to mix) or from the routing mix (5f). So be careful not to choose both, it will increase the outgoing level by approximately 6dB.

LEDBAR

The 6 segment ledbar (the first led only indicates that the channel receives its powersupply) is a peakreading instrument conform world standards according attack and release times. It reads the outgoing level which is on the output-jack of the channels backplate. This could be the channel postfader signal or the groupoutput signal (if you have pushed one of the 1 a-b-c-d switches, read 2.1 CHANNEL OUTPUT SECTION).

2. CHANNEL INPUT SECTION

The channel can operate in either the microphone or line input modes.
The microphone input is an electronically balanced, transformerless design. The input-impedance is greater than 2 Kohms, which will not cause any loading effects on todays studio microphones.

The line level input which is also to be used as effect and tape return has an input impedance greater than 10 Kohms, which is high enough to interface with all available peripheral equipment.

2.a +48 VOLT

The +48 Volt switch is there to feed condensor microphones and Direct Injection boxes (if they have that facility). When using other microphones such as dynamic and electret ones, the phantom power supply should not be switched on.

2.b PAD

Pushing the pad switch inserts a 20dB attenuation into the input of the microphone amp. This could be necessary when modern capacitor microphones are to be used in close proximity to musical instruments. Even D.I. boxes are capable of providing high level signals.
The pad switch also raises the input impedance providing a balanced line input when the need arises. In special cases we can increase the pad to 30dB and the impedance too, to provide for balanced switched mic/line inputs.

2.c MIC/LINE GAIN

The microphone input can be varied between 20dB and 55dB of gain. The pad of 20dB increases the control range to 55dB.The line input gain can be varied between -20dB and +20dB.
Both the mic and line amplifiers have their own input connectors. The mic amp is balanced on XLR. (1 is earth, 2 is hot in phase and 3 is cold out of phase)
The line amp is unbalanced on a stereo jack. Tip is hot, ring is earth and shield is earth too.

2.d. PHASE

The phase reverse switch changes the wiring of the mic input only. In most cases it is the mic signal that's out of phase with another source.

2.e. LINE/GROUP

The line input is selected by pushing the line/group button. The line input can also amplify the group output when an external patch is made between the channel output and the line input. A set up to create equalized submasters.

3. EQUALIZER SECTION

The equalizer section of the DAYNER stands out by its very effective design. It allows 4 sections of control over the entire audio spectrum.

3.a. HIGH

16dB of boost or cut is available at 10kHz, with a shelving curve, which means that when the desired amount of boost or cut is reached the curve shelves from that frequency on.

3.b. FREQUENCY (MID 1)

This control selects the centre-frequency of the mid 1 section. It range from 1kHz to 10kHz.

3.c. MID 1

This control has a range of + and -16dB with a "bell" curve. Having reached its maximum/minimum at the selected frequency (control 3.1.2) the amplitude response returns to zero on either side of that selected frequency.
A plot from that response shows a bell shape. The bandwidth of that bell curve is fixed at 1,5.

3.d. FREQUENCY (MID 2)

This control selects the centre-frequency of the mid 2 band. It ranges from 100Hz to 1kHz.

3.e. MID 2

This controls is exactly doing the same as the mid 1 control (3.1.3) but now for the selected frequencies by the mid 2 frequency control.

3.f. LOW

The low control has a shelving characteristic just like the high control.
16dB of boost or cut is available at 60Hz.

3.g. EQ-ON

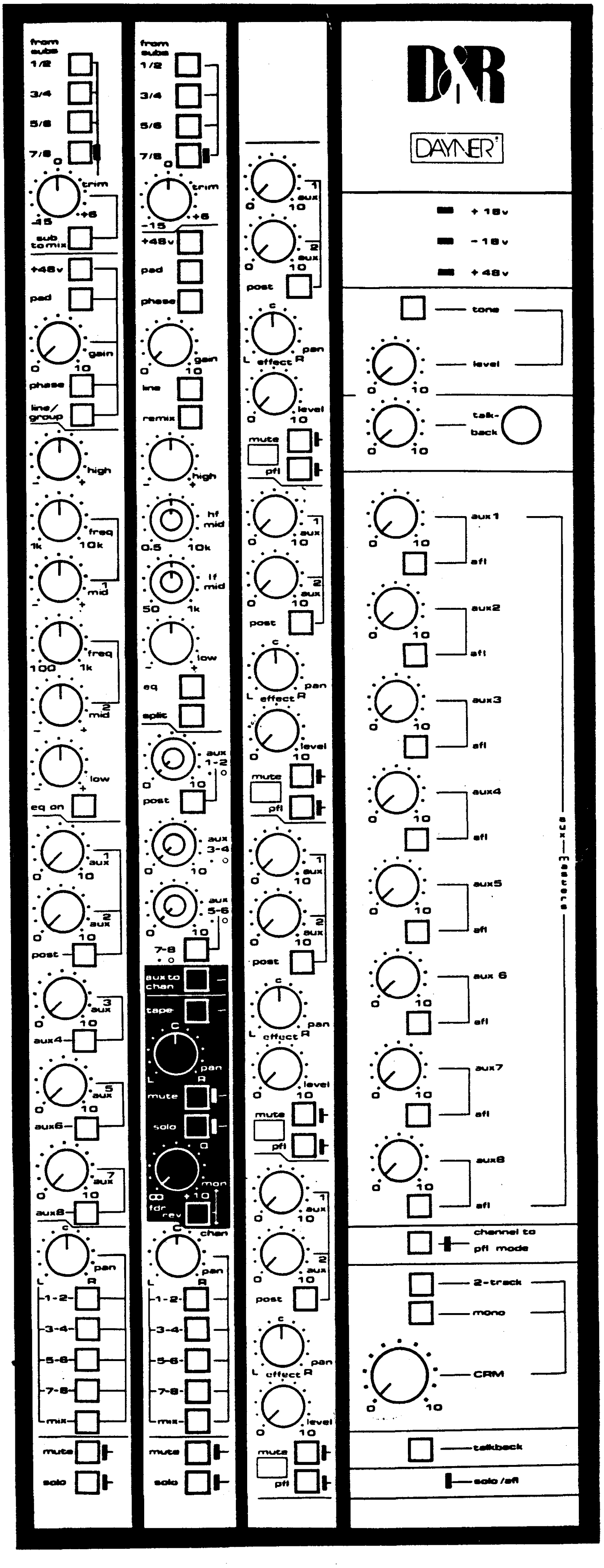
The equalizer section can be switched in or out for comparing equalized and non equalized signals.

4. AUXILIARY SECTION

There are 5 aux sends controls available and 8 aux busses in total. This seems quite a bit, but to days extensive signal processing requires a lot of aux sends.

4.a.b.c SENDS 1 AND 2

Auxiliary sends 1 and 2 are normally pre-fader but can be switched post-fader if desired. They are intended to be used as stereo foldback sends during recording/public address set-ups and switched post-fader during remix sessions. The aux sends are wired post equalizer, post insertion point and channel mute switch.



DAYNER
+18v
-18v
+48v
tone
level
talk-back
aux1
afl
aux2
aux3
aux4
aux5
aux 6
aux7
aux8
channel to pfl mode
2-track
mono
CRM
talkback
solo/afl
from subs
1/2
3/4
5/6
7/8
trim
sub to mix
+48v
pad
phase
gain
line/group
line
remix
high
freq
mid
hf mid
lf mid
low
eq on
eq
split
aux
post
aux to chan
tape
pan
mute
solo
mon
fdr rev
chan
1-2
3-4
5-6
7-8
mix
effect
pfl

CHANNEL FADER

The channel fader has a slide length of 100 mm and is manufactured to give an exceptionally smooth feel in operation.

7. CHANNEL IN/OUTPUTS

7.a. This is the XLR input for balanced condensor or dynamic microphones. Pin 1 is earth, pin 2 is in phase (hot) and pin 3 is out of phase (cold).

7.b. This is the line input, which is unbalanced. The tip is hot while the ring and sleeve are wired to earth. This input has a sensitivity of - 20dB maximum to infinity. The input impedance of 10Kohm will not load any line output of tape recorders or signal processors.

7.c. This is the output of the channel. The tip has a nominal level of -10dBV and the ring + 4dBu.
One of the two level outputs has to be left unconnected. The - 10 dBV is for semi pro equipment and the + 4dBu for pro equipment. The output signal can be post channel fader or the group output when a from sub switch is activated.

7.d. This is the channel insert (immediately preceding the channel fader). The tip is the return, while the ring is the send.
In/out level is 0dBu.

7.e. This is the group insert. The tip is the return and the ring the send. This insert can also be used as an effect input with the trimpot as level pot and the mix push-button in the output section as a way of routing the effect to the master.
Odd channels to the left, even channels to the right.

8. DAYNER IN-LINE MODULE

Switches and controls

8.a. **From subs**: used with the floating subgroup system to assign which output you want your subgroup to assign to. Module #1 would be output #1 to the input or track #1 on your multitrack and module #24 would be the output #24 to the input of track #24 on your multitrack. Black switch caps are odd (pan left) and grey switch caps are even (pan right) on the channel assign panpots, just above the channel faders.

8.b. **Trim**: Output trim (volume) of the summed output on the same channel. It can also be used as a subgroup volume.

9. **48V**: Phantom power switch for condenser microphones.

10. **Pad**: Using this switch inserts a 20dB attenuation into the mic input amp. If the signal source is too loud, you use this in conjunction with the mic/line gain to allow you more control on the channel faders.

11. **Phase**: Use this switch to reverse the phase of any mic input coming from any mic or signal that may be out of phase with other mics or signals. A good way to check for "out of phase" is to push the mono switch on the master section and listen real close to the mix. If you hear something that sounds strange or completely missing in your mix, push the phase switch on those channels suspected. If the sound returns or sounds better, that channel was out of phase with the others. You can have an acoustical phase cancelation as well when using multiple mics on the same signal such as drums, vocals, horns, strings, etc.

12. **Gain**: This is the single most important control on your console. When this control set properly, you can achieve the very best signal to noise ratio and get the most headroom needed for high quality recordings. After plugging in a mic, push the "channel to pfl mode" switch on the master section. Now push the solo switch just above the channel fader on the channel you're setting, with the channel fader off. Turn the gain control clockwise until you see a "0" output level on the master meters. Now slide up the channel fader to "0". Remember, if the signal sorce gets louder or softer, you may have to go back and check this setting. The volume can also change if you boost or cut in the equalizer. Be sure the signal being miked stays the same volume when you start recording, or you'll need to go back and do all this again. Do this with every microphone input to achieve the high quality sound D&R products are known for.

13. **Line**: Switches from mic input to line input on the channel.

14. **Remix**: Switches tape return input to channel input (controlled by channel fader). At the same time switching the line input to the monitor section controlled by monitor pot.

15.a. **High**: Boost or cut 16dB at 10.000 Hz shelving (boost or cut the same amount above 10.000 Hz).

15.b. **HF MID**: Boost or cut 16dB bell curve sweepable from 500 Hz to 10.000 Hz with a width of 1.5 octaves. Top control is boost or cut and bottom control is the frequency sweep.

15.c. **LF MID**: Boost or cut 16dB bell curve sweepable from 500 hz to 1.000 hz with a width of 1.5 octaves. Top control is boost or cut and bottom control is the frequency sweep.

15.d. **LOW**: Boost or cut 16dB at 60 Hz shelving (boost or cut the same amount below 60 Hz).

15.e. **EQ**: Switches equalizer in or out of the circuit.

16. **SPLIT**: Pushing this switch allows you to actually split the equalizer into two sections. The high and low shelving sector will switch to the monitor and the two sweepable mids will switch to the channel allowing you to have equalization on both inputs on each module. Eq switch must be in down position.

17.a.b **AUX 1-2**: Top control is aux send 1 and bottom control is aux 2. This pair of aux sends are most commonly used for cue sends (stereo headphones), top control being left or cue 1 and bottom control being right or cue 2. These aux sends can be effect sends in the mixing session.

17.c. **POST**: This switch allows you to switch where aux 1 and 2 get their signal from. In the up position (pre) the signal comes from before the fader and in the down position (post) the signal comes from after the fader. It´s a good idea to use the pre position when using aux 1 and 2 for cue or headphones so when changing the monitor levels you won´t affect the headphone mix.

17.d.e **AUX 3-4**: Can be used as effect sends or a seperate mix for monitoring.

17.f.g **AUX 5-6**: Same as 3-4.

17.h. **AUX 7-8**: This switch allows you to switch the bottom concentric pots from being aux 5-6 to become 7-8.

18. **AUX TO CHANNEL**: This switch will allow the aux sends to get their signal from the channel fader or from the monitor pot.

19. **TAPE**: This switch will switch where the monitor gets it´s signal from. In the up position the channel fader feeds the monitor and in the down position the tape return feeds the monitor.

20. **PAN**: This control will pan the signal in the monitor section between the stereo image on the mix buss.

21. **MUTE**: This switch will turn the monitor off on that module.

22. **SOLO**: This switch will allow you to hear only the signal in the monitor section. Depending upon what position the "channel to pfl mode" switch (located in the master section) is in, you may hear the signal mono pre fade listen or stereo in place following the panpot.

23. **MON**: This controls the volume in the monitor section of that channel. It gets it's signal depending on what position the tape switch is in.

24. **FDR/REV**: This switch will allow you to swap places with the channel fader and the monitor pot. In the down position, the channel fader controls the monitor section and the monitor pot controls the output of the channel to the subgroups or summed output. By putting the monitor section on the channel faders, those producers can play with the faders without affecting the signal going to the tape machine.

25. **CHANNEL PAN**: Allows you to place the signal anywhere in the stereo image, feeding the mix buss or the subgroups.

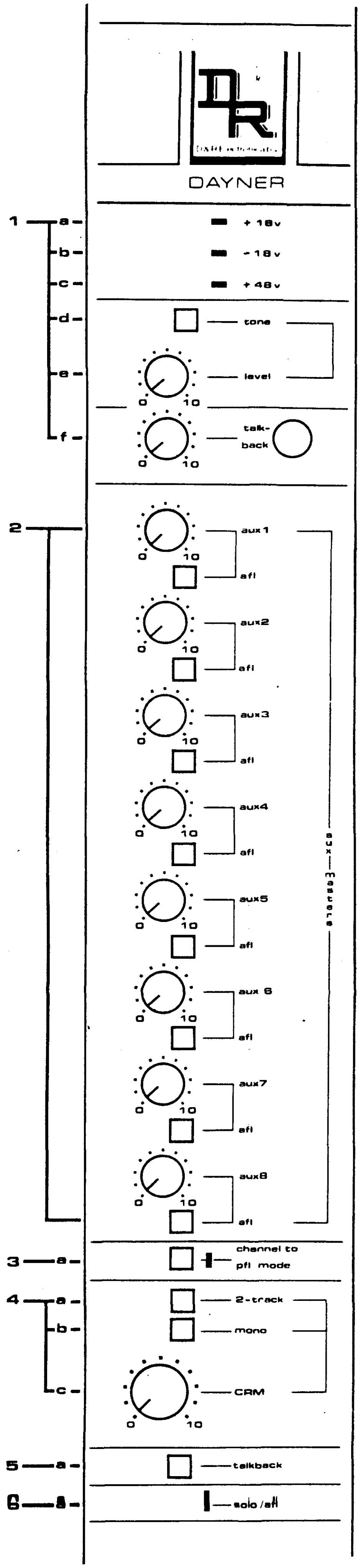
26.a. 1-2: Assignment switches to subgroups 1 & 2.
26.b. 3-4: Assignment switches to subgroups 3 & 4.
26.c. 5-6: Assignment switches to subgroups 5 & 6.
26.d. 7-8: Assignment switches to subgroups 7 & 8.

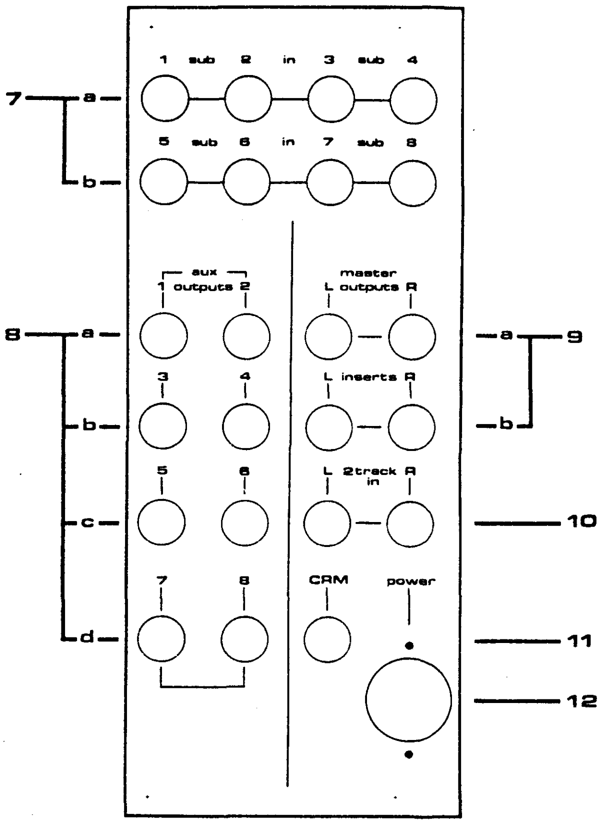
27. **MIX**: Assigns channel to stereo mix buss.

28. **MUTE**: Turns off the channel.

29. **SOLO**: Allows you to hear the channel only on your monitor speakers.

MASTER MODULE





MASTER MODULE

The master module of the DAYNER contains all the electronics for the summing of the left/right signals, the aux signals, the subgroup amps as well as the Control Room Monitor section.
The width of this module is 90 mm. (3 times the channel module / blind panel / patch panel / effect / tape / return module).

1.a.b.c. POWER SUPPLY STATUS

These three status leds indicate that power has been send to the DAYNER. +/- 18 Volt for the electronics and + 48 Volt for the phantom powering of condenser microphones and D.I. boxes.

1.d.e. TONE GENERATOR

The DAYNER has a low distortion phase shift type oscillator which produces a 1kHz sine-wave.
The tone can be adjusted in level from zero to about + 10dB. Push-button switch 1 d routes the oscillator signal to <u>all</u> mix busses in the DAYNER. To provide the channel output with the 1kHz sine-wave it is necessary to activate one of the from sub switches. The trim control is there for fine adjustment per channel. All other outputs, such as Aux outputs and master left/right outputs <u>only receive</u> the 1kHz tone if their <u>associated master controls are opened</u>. The tone switch also dims the C.R.M. output by 20dB.
A convenient feature.

1.f TALKBACK MIC

The built in electret microphone can be adjusted in level and is permanently routed to the Aux 1 and 2 busses. To avoid feedback the talkback switch 5a also dims the C.R.M. output by 20dB.

2. AUXILIARY MASTERS 1-8

These 8 controls handle the outgoing levels to the 8 aux outputs.
Every output has an associated A.F.L. (after fade listen) button to allow the signal to be monitored and metered.

The nominal outgoing level is + 4dBu. In P.A. situations only 600 ohm headphones are to be connected. If overall level is too low, contact your dealer for a small modification.

5.a. TALKBACK SECTION

This switch routes the talkback to the Aux 1 and 2 busses. At the same time it dims the C.R.M. output by 20dB to avoid feedback. An electronic delay in the activating of the electret microphone avoids mechanical noise when the T.B. switch is pushed. Routing to aux 3-8 is optional.

6.a SOLO/A.F.L.LED

If anywhere in the console a solo/a.f.l. button is activated, this led lights. You hear the selected signal instead of the usual master mix.

7. MASTER IN/OUTPUTS

This part of the DAYNER provides all the connections with the external equipment, such as signal processors and power amps.

7.a.b. These 8 sub inputs directly accept 0dBu level signals. These inputs are directly wired to the summing amps of the 8 subgroup amplifiers. They are intended to be used as effect returns, as tape returns or as universal line inputs. Think of the "sub in" inputs as 8 extra line inputs. To bring the signal of, for instance, sub in 1 into the master you only have to push a from sub 1/2 somewhere in the console and then route it to the master by the sub to mix control in the channel output section.
The odd numbered sub-in signals will appear in the left master output and even numbered ones in the right master output. The input impedance is 10Kohm.

8.a t/m d AUXILIARY OUTPUTS

These 8 outputs allow connection to the foldback amps and/or signal processors.
The outgoing level is + 4dBu on tip. The ring is connected to earth.

9. MASTERS

9.a These 2 jacks are devoted to the master left/right insert. The level is ØdBu. The ring is send, the tip is return. These 2 inserts are provided to insert limiters or other devices into the stereo mix immediately before the master faders.

9.b These are the master left/right outputs of the DAYNER. Signal level is a nominal + 4dBu on the ring. The tip is wired to - 1ØdBV for easy interfacing of semi pro equipment.

1Ø 2 TRACK

These 2 jacks accept external stereo sources such as 2 track machines. These inputs also have two input sensitivities, - 1ØdBV on the tip and + 4dBu on the ring. The DAYNER easily interfaces with to days two-track master machines.

11. C.R.M.

This is the stereo output of the Control Room Monitor. Nominal level is around + 4dBu. Tip is left and ring is right. Do not load this output with 8 ohm headphones but only with <u>6ØØ ohm</u> ones.

12. POWER

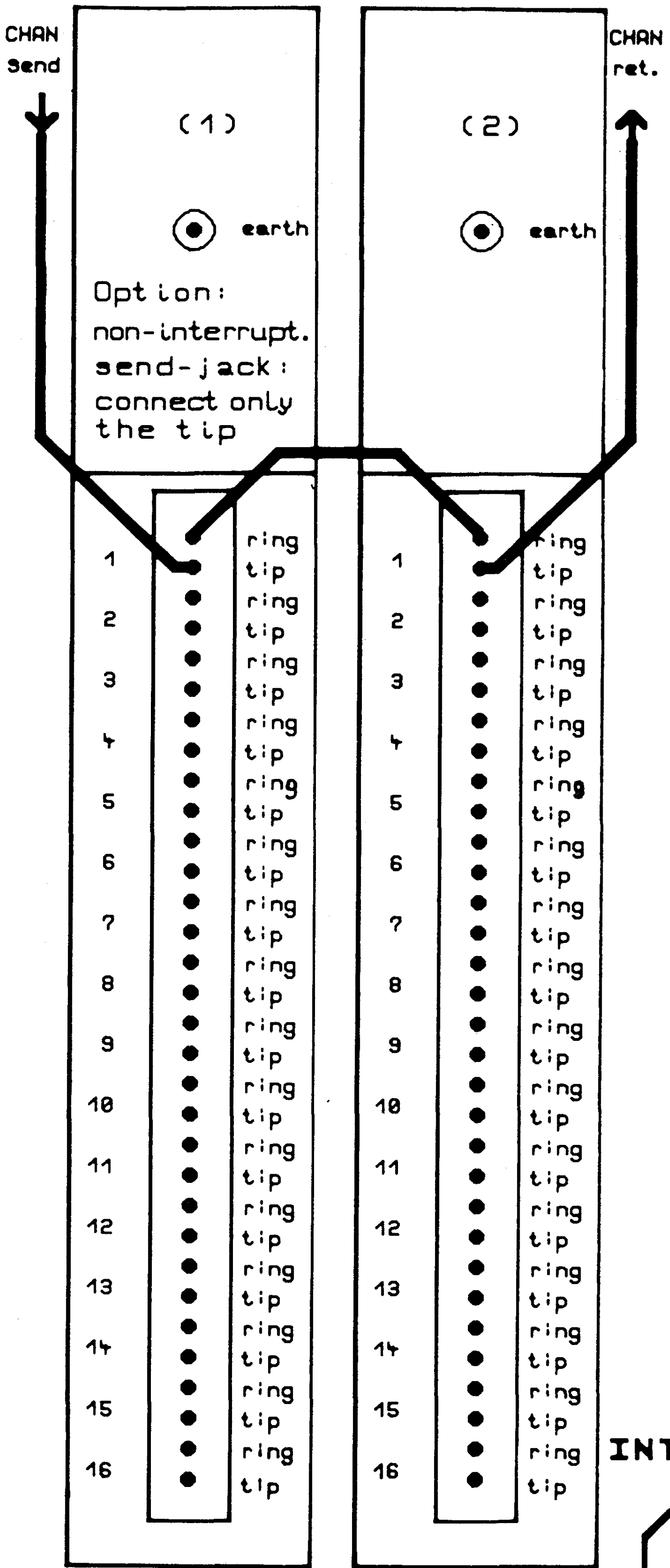
The DAYNER is powered from an external heavy duty power supply. The supply voltages of +/- 18 Volt and + 48 Volt are fused in the power supply. The master section indicates by leds if the voltages are present. The power supply is connected by way of a 5 pin XLR type of connector.

PATCH PANEL

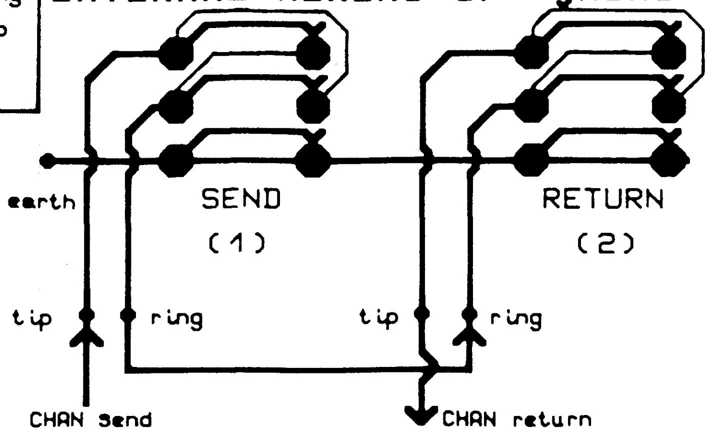
The Patchpanel modules of the DAYNER contain 16 breakjacks per module. These are wired to a 32 pin Molex connector on the back. The way of external wiring allows any desired configuration.
Simple in/out in one jack, inserts in one jack with break contacts, seperate send returns between jack 1 and 2 or between jack 1 from module 1 and 1 from module 2.
The break contact wiring is designed in a way that always the break contacts are paralled. Avoiding noise and interruption after extensive use.
The Molex connectors on the back are from the male type. The female connectors have to be wired permanently in a way your situation desires.
The 32 pins are wired to the tip and the ring of the 16 jacks, while the break contacts of the jacks are permanently shorted.
To use jack 1 as an ordinary channel insert. Connect the tip to the tip of a channel insert and the ring to the ring of that channel insert. Now you have moved the channel insert from the back of the console to the easily accessible jack of the patch module.
Many configurations can be realized in this way. Before we show you in easily understandable drawings how to realize some configurations it is necessary to know how to earth the patch panel jacks. On the back of the patch module you will find an earth tag. This has to be wired externally to the master earth tag (on the back of the master module).
Every connection from the channel to the patch module has to be earthed only on the channel side. Cut the shield on the patch module side and only connect the wire (or wires) to the tip and/or ring of the associated patchjack.
External equipment has to be earthed on the patch earth-tag. It is wise to wire all the external equipment inputs to one patch panel and the output to an adjacent module. Now connect all the shieldings to the source earth of the equipment.
Always connect one side of the shielding only. We will later explain more about wiring and shielding.

DAYNER-PATCH PANEL

Example of patch connections.



INTERNAL WIRING OF JACKS



OPERATION

INTRODUCTION:

The DAYNER is designed to be the perfect answer for a 2 to 24 track recording studio.
To get more familiar with the DAYNER we shall discuss the whole recording process and divide it into 4 basic sequences:

1 RECORD

Recording from microphone/line input on to the multitrack. This could be from one or more channels at the time.

2 MULTITRACK REPLAY (SYNC)

In this mode you listen to what has been recorded.

3 OVERDUBBING

This means listening to already recorded tracks and recording on spare tracks, until all tracks are filled up with music.

4 REMIX

Replaying of all recorded tracks together with signal processing reverb and all that is necessary to create the final mixdown.

1 RECORD MODE

This is the beginning of a session.
All input channels are placed in the mic mode by leaving the line/group switch in the up position. Phantom powering is applied if necessary. Eq. switch in the up position.
The signal flows through the fader and is available postfader, at the back of the channel, at the output.
The led bar reads the outgoing signal.

MONITORING

The recorded signal can be monitored in various ways. The DAYNER design is intended to be used as a console with channels for every signal source. So the easiest way is to monitor the tape output in another channel. But in low cost configurations it is possible to monitor through the channel output section.
Bring the tape in/output by a stereo type jack into the group insert jack. Connect the tape output to the tip of the jack and connect the ring to the input of the tape-recorder. The in/outgoing level is restricted to a nominal level of ØdB.
Although the fader will determine the recorder level. The trim control sets the replay level and the mix button determines whether the replay signals are to be heard in the left or right output. This is a restriction, we know.
But it is also not the best way to use the DAYNER. In this way 8 channels are sufficient for 8 track recording.
A third way of using the DAYNER is to connect your tape deck between the line input and output of the same channel. Now you can record from the mic input and replay from the line input
(Note! : switch your eq. off while replaying).
There is one thing to be noted. When your tape deck gives its input signal at its output you have a loop. From line in to direct out.
Use a from sub switch in that channel to interrupt this feedback loop.

EFFECT RETURN MODULE

The fourth way of monitoring your tape signal is by way of the effect/return module. This module contains 4 separate returns in one module. Every return input goes via a level control and pan-pot to the master mix buss.
P.F.L. and effect sends are available too.
In this way 8 tracks monitoring with only two modules is available.
A typical set up for an 8 track recording session is to connect the in/outputs of the 8 track machine. Now you can record into these 8 channels from left over channels.
You decide how many you need and route them by way of F.S.S. (system).
Effects can be recorded via the 8 sub in jacks. If the channel signal is led through the sub amps the effect signal can be mixed in.

The various gain and level controls throughout the signal chain may be adjusted for optimum signal to noise ratio.

MULTIPLE SOURCES ON 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 will be required.
This can be done easily on the DAYNER by way of the internal floating subgroup amplifiers located on the mastersection p.c.b.'s.
Simply route to one of the 8 subgroups by pushing one or more of the number 5 section channel routing switches. Decide on which track you want to record these signals and activate the related "from sub" switches in these channels. The channel metering will show the subgroup level which can be overall changed by the trim controls. To monitor this subgroup pre-tape, activate the group mix routing switch. To monitor post tape, use another channel, or wire up your system for the budget way of multitracking. Now your trim control gets another function. Carefully follow the signal flow diagram to understand all these explanations.

MICROPHONE GAIN

The amount of gain required depends on the type of microphone, the sound pressure level and the distance between the sound source and microphone.
A 20 dB pad can be inserted where levels are too high.

INSERT CHANNEL/GROUP

If the dynamics are too high a compressor/limiter can be inserted in the channel or even in the group insert, if a whole group signal has to be compressed. This all depends on the situation.

FOLDBACK/EFFECT SENDS

During recording it is essential that every one hears what's going on. Headphone mixes are usually derived from pre fader auxiliaries.
In the DAYNER Aux 1 and 2 are ideal for this purpose, especially while the talkback is routed to these busses. The best way is to derive the Aux sends from monitor channels. This set up is very useful for dubbing situations. In the budget set up it is not possible to monitor post tape. A suggestion for the musicians is to monitor from the main outputs. By using the effect return modules monitoring is possible of course.

EFFECT SENDS

All unused aux sends can be used to send signals to signal processors, such as reverb and delay. The Aux sends are usually post fader to always keep the right balance between untreated and treated signals.

EFFECT RETURN

To bring the effect on tape it has to be mixed with the original. Replay the effect in an unused channel and route it to the recording channel by the routing buttons and the from sub switches. Also send the source original through the subgroup system. In the budget configuration the sub-in jacks can be used to replay the effects or the unused line inputs from already recorded tracks. Pushing the line/group switch, will bring the tape machine in the channel (see signal flow diagram).
If you have chosen for extra effect/return modules, your effect can be routed only to the masters via these modules.

2 MULTITRACK REPLAY

This is already discussed in section 1 (record mode) but replaying is normally done through channels and in a budget version through the group insert, or even through the sub-in jacks (8 maximum). The effect/return modules can be used as tape replay also of course.

3 OVERDUBBING

Overdubbing is the process of building up a recording track by track, while listening to previously recorded tracks. Some channels will be in the microphone mode while others are replaying the multitrack .
The replaying tracks are monitored by separate channels for instance left of the master section, while the recording channels are to right of the master section and routed to the outputs of the leftside channels.
Headphone monitoring can be best mixed from the replay channels. Here a decision has to be made as to the source of the headphone mix. It can be derived from the input channel, replay channel or both. It is up to you.

From the input channel the musician will hear himself only, not the previous recorded signal on that track. A most convenient way of headphone mix today is as follows. Most of todays multitrack will give the input signal at its output when in recording.
Derive the headphone mix from the tape. As long as the tape is in the replay mode the musician will hear his previous recorded tracks and the moment the engineer goes into record he will hear himself life. This method saves the engineer from continually switching monitor sources.
By adding the aux signal from the recording channel the musician will hear himself before the moment of recording too. As soon as the multitrack goes into recording a slight increase in the level will be heard. Not too bad, because the musician exactly knows that what he is doing will be recorded.

4 REMIX

Remix is the process of combining all recorded track together with extensive signal processing.
In the DAYNER, in its dual mode, all tape tracks are already in the remix mode. It is only necessary to switch the microphone inputs to line in the recording channels to have all the effect outputs available, which are wired to the recording channel line inputs. All incoming signals can be routed to the stereo master through the mix push-button. Subgroups can be made as desired in the same way as during recording.
The Aux sends 1 and 2 can be switched to post if necessary.

P.A. SET UP

The DAYNER is an ideal console for P.A.. It is so compact even with a lot of channels in use. In P.A. situations it is often desired to have permanent subgroups. This can be accomplished in two ways.
1- without loosing input channels
2- with separate output channels with all facilities.

1 SUBGROUPING WITH LIMITED NUMBER OF AVAILABLE CHANNELS

Route the input signal in the normal way to the 8 subgroups. Then activate in channel 1 and 2 the "from sub 1/2" switch together with the mix switch. In channel 3 and 4 the "from sub 3/4" and the mix switches. For channels 5 to 8 in the same way.
The trim controls will set the overall level of the subgroups in pairs to the master mix.

In this way you have all input channels available for signal sources while 8 output sections <u>any where</u> in the console can be used as subgroup masters. The outputs of the channels where the subgrouping is realized will give the group signal for recording or any stereo mix you like out of the individual channel. It is also possible to normally route to the master from the channels and at the same time to have 4 stereo mixes for other purposes.
I can think of a mix without any leadvocals for dubbing later on. I´m sure you have already more possibilities in mind.

2 SUBGROUPING WITH EXTRA CHANNELS

The DAYNER has the facility to change an input channel into an output channel.
Just route in the normal way and activate the "from sub" switches in unused channels. Patch the output in this "from sub" channel to the line input. Now activate the line/group switch and route the incoming group signals to the master mix.
<u>Never activate the same numbered group switches in these channels as you have done in the output section of that channel.</u> Feedback will be the result, see your flow chart.
In this way you can have 8 separate output channels with full equalizing, aux sends and faders. A very convenient set up for P.A..
To indicate these submaster channels easily, we have supplied you with 8 extra fader knobs, in a different colour. Slowly draw the knobs from the shaft and push the new ones on the shaft in those channels where you want a different colour coding.

CONNECTORS

<u>Channels:</u>

XLR inputs:	level	:	-70 to -20dB
	pin 1	:	signal ground.(screen)
	pin 2	:	signal high (in phase, hot)
	pin 3	:	signal low (out of phase, cold)
Line inputs:	level	:	-20 to infinity
	tip	:	signal high (in phase, hot)
	ring	:	signal ground
	sleeve	:	signal ground

Insert: channel/group	level	0dB
	tip	return signal
	ring	send signal
	sleeve	ground
Outputs:	tip	-10dBv
	ring	+4dBu
	sleeve	signal ground

OUTPUTSECTION

Sub in:	level	0dB
	tip	signal high (in phase, hot)
	ring	signal ground
	sleeve	signal ground
Aux outputs:	level	+4dBu
	tip	signal high (in phase, hot)
	ring	signal ground
	sleeve	signal ground
Inserts:	level	0dB
	tip	return signal
	ring	send signal
	sleeve	signal ground
Master outputs left/right:	level	-10/+4
	tip	-10dBV (300mV)
	ring	+4dBU (1.22 Volt)
	sleeve	signal ground
2 Track:	level	-10/+4
	tip	-10dBV (300mV)
	ring	+4dBU (1.22 Volt)
	sleeve	signal ground
C.R.M :	level	+4dBU (1.22 Volt)
	tip	left output
	ring	right output
	sleeve	ground
Power (XLR):	pin 1	ground
	pin 2	+18 Volt
	pin 3	-18 Volt
	pin 4	phantom
	pin 5	chassis ground

Typical shielding situations:

Output	Input	Screen
Unbalanced	Unbalanced	Source
Unbalanced	Balanced	Source
Unbalanced	Differential	Source
Balanced	Unbalanced	Destination
Balanced	Balanced	Source
Balanced	Differential	Destination
Differential	Unbalanced	Source
Differential	Balanced	Source
Differential	Differential	Source.

Balanced means transformer balanced, while differential is electronically balanced.
There are some cases which give better results in practise. Always connect one at the time and check.
Always use twin screened audio cables and connect both conductors at both ends, the shielding at one and. (except patch, cords, these earths are tied together in the console).
We know that this part is a difficult one but once properly installed and wired, the results will be clean and noise free.

FAULT FINDING

It is essential to study the signal flow chart carefully. Only in this way you can isolate problems in the DAYNER. By following the signal through in and output jacks its is possible to locate a fault. If a fault is located, inform your dealer or us and we will try to help you by advice if this will not help just return the channel or master to your dealer or the factory and we will be happy to repair it within 24 hours.
Many faults can be found by logical thinking and replacing integrated circuits, which is very easy they are all socketed.

REMOVING A MODULE

Switch off the power supply first.
Remove the 2 module retaining screws, which will allow to carefully withdraw the module from the console. First lift the fader side of the module, remove the flatcable connector and then further upwards. Then remove the second flatcable connector and the 3 pin connector from the XLR input. Now extender cables can be connected (if ordered).
The master section can be lifted in the same way, but we advice to service the master section only by qualified personal.
The patch panel has no flatcable wiring underneath.

In this manual we have tried to give you an oversight of all the possibilities the DAYNER offers you. If there are questions left do not hesitate to contact us or your dealer.
With the DAYNER we are sure there is no limit to your creativity anymore.
We wish you many years of enjoyable music.

Best regards,

D. de Rijk
president.