EURODESK MX8000A

User´s Manual

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by Wilf Smarties

Version 1.0 1996

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CE

EG-Declaration of Conformity

BEHRINGER®
INTERNATIONAL GmbH

acc. to the Directives 89/336/EWG and 73/23/EWG

We, **BEHRINGER INTERNATIONAL GmbH**
Hanns-Martin-Schleyer-Straße 36-38
D - 47877 Willich

Name and address of the manufacturer or the introducer of the product on the market who is established in the EC

herewith take the sole responsibility to confirm that the product:

EURODESK MX8000A

Type designation and, if applicable, Article-Nº

which refers to this declaration, is in accordance with the following standards or standardized documents:

☒ **EN 60065**	☒ **EN 61000-3-2**
☒ **EN 55020**	☒ **EN 61000-3-3**
☒ **EN 55013**	

The following operation conditions and installation arrangements have to be presumed:

acc. to Operating Manual

BEHRINGER
INTERNATIONAL GmbH
Hanns-Martin-Schleyer-Str. 36-38
D-47877 Willich-Münchheide II
Tel.-Nr. 0 21 54 / 92 06-0
Fax-Nr. 0 21 54 / 92 06-30

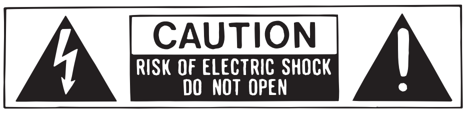
B. Nier, President — Willich, 01.05.1999

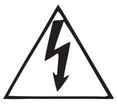
Name, address, date and legally binding signature of the person responsible

SAFETY INSTRUCTIONS

CAUTION: To reduce the risk of electrical shock, do not remove the cover (or back). No user serviceable parts inside; refer servicing to qualified personnel.

WARNING: To reduce the risk of fire or electrical shock, do not expose this appliance to rain or moisture.





This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure - voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

DETAILED SAFETY INSTRUCTIONS:
All the safety and operation instructions should be read before the appliance is operated.

Retain Instructions:
The safety and operating instructions should be retained for future reference.

Heed Warnings:
All warnings on the appliance and in the operating instructions should be adhered to.

Follow instructions:
All operation and user instructions should be followed.

Water and Moisture:
The appliance should not be used near water (e.g. near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool etc.).

Ventilation:
The appliance should be situated so that its location or position does not interfere with its proper ventilaton. For example, the appliance should not be situated on a bed, sofa rug, or similar surface that may block the ventilation openings: or placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

Heat:
The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliance (including amplifiers) that produce heat.

Power Source:
The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

Grounding or Polarization:
Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

Power-Cord Protection:
Power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles and the point where they exit from the appliance.

Cleaning:
The appliance should be cleaned only as recommended by the manufacturer.

Non-use Periods:
The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

Object and Liquid Entry:
Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

Damage Requiring Service:
The appliance should be serviced by qualified service personnel when:
- The power supply cord or the plug has been damaged; or
- Objects have fallen, or liquid has been spilled into the appliance; or
- The appliance has been exposed to rain; or
- The appliance does not appear to operate normally or exhibits a marked change in performance; or
- The appliance has been dropped, or the enclosure damaged.

Servicing:
The user should not attempt to service the appliance beyond that is described in the Operating Instructions. All other servicing should be referred to qualifield service personnel.

FOREWORD

Dear Customer,

Welcome to the team of EURODESK users and thank you very much for expressing your confidence in BEHRINGER products by purchasing this unit.

It is one of my most pleasant tasks to write this letter to you, because it is the culmination of many months of hard work delivered by our engineering team to reach a very ambitious goal: making an outstanding device that will become a standard tool used by studios and P.A. companies. The task to design the EURODESK certainly meant a great deal of responsibility, which we assumed by focusing on you, the discerning user and musician. It also meant a lot of work and night shifts to accomplish this goal. But it was fun, too. Developing a product usually brings a lot of people together, and what a great feeling it is when everybody who participated in such a project can be proud of what we've achieved.

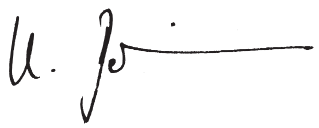
It is our philosophy to share our joy with you, because you are the most important member of the BEHRINGER family. With your highly competent suggestions for new products you've greatly contributed to shaping our company and making it successful. In return, we guarantee you uncompromising quality (manufactured under ISO9000 certified management system) as well as excellent technical and audio properties at an extremely favorable price. All of this will enable you to fully unfold your creativity without being hampered by budget constraints.

We are often asked how we can make it to produce such high-grade devices at such unbelievably low prices. The answer is quite simple: it's you, our customers! Many satisfied customers means large sales volumes enabling us to get better conditions of purchase for components, etc. Isn't it only fair to pass this benefit back to you? Because we know that your success is our success, too!

I would like to thank all people whose help on "Project EURODESK" has made it all possible. Everybody has made very personal contributions, starting from the designers of the unit via the many staff members in our company to you, the user of BEHRINGER products.

My friends, it's been worth the trouble!

Thank you very much,



Uli Behringer

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1. THE MANUAL

1.1 NOMENCLATURE

Most specialist subjects are not really all that difficult provided you understand the LANGUAGE USED, and the vocabulary of mixing is pretty straight-forward. Nevertheless it is as well to be clear about what certain terms mean. A ‘slot’ in a recorder will always be referred to as a TRACK, while that in a mixer will invariably be a CHANNEL. A GROUP will always refer to a sub-mix of channels, never a collection of musicians. Similarly the term BAND will be mentioned only in conjunction with FREQUENCY. I will attempt to be as unambiguous as possible with terms, since much confusion can arise from sloppy definitions. If a new term (to you) does arise, and it isn’t fully explained in the text, look it up in the GLOSSARY before you write me a letter of complaint!

Some terms, like love, can have a plurality of meanings. TRACK, in mixing parlance, refers to a tape recorder. In electronic circuits, components on a PCB (printed circuit board) are linked by flat conductors called TRACKS. Hopefully, where terms have different meanings the contexts will be sufficiently diverse so as to avoid any possible confusion.

1.2 AN UN- HOLISTIC APPROACH

We all know about the interconnectedness ufl everything. A butterfly’s wing causes a hurricane half-way ‘round the world etc.. It is virtually impossible to fully explain one aspect of a mixing console-(e.gj CHANNEL ROUTING) without also making it clear what those routes are, where they go, are they migratory etc. etc.. I have compartmentalised the EURODESK manual into Sections, making it easy to find problem solving information and advice. You might find that several (though not too many I hope) cross-sectional references have been made, where areas of interest overlap. E.g.: Channel EQ is specified and described in the CHANNEL Section 3.3, while EQUALISATION has it’s own section, reflecting it’s importance and weight as a subject in it’s own right.

P.S.: If I keep repeating myself concerning the use of B-Channels and the MIX-B bus, it’s because a proper understanding of this area of the board will greatly expand your mixing repertoire.

1.3 KEY

All DESK functions will be numbered consistently throughout the manual, whether they be in the text or in an illustration. In addition the following prefixes will be used to denote the various types of function control in any illustrations !\ text respectively:

S for Switch,

L for LED,

P for Potentiometer and

F for Fader.

After every“ prefix you will find the FUNCTION NUMBER. Numbering starts at the top of a CHANNEL, works its way through a stereo GROUP, and finally through the MASTER SECTION. The phantom power and tape operating level switches are not included in the numbering system.

2. EURODESK OVERVIEW

2.1 ARCHITECTURE

The EURODESKI (fig. 2.1, page 2-2) is a hybrid SPLIT /I INLINE console. Input channels cover most of the surface from the left, while the outputs to tape are to the right. Tape monitor returns, however, are housed within the channel strips, not above the tape outputs, as would be the case in a conventional ‘split’ design. This architecture enables much flexibility to be bestowed onto the tape monitor signal path, not least being its ability to pick up functions easily from the main channel. Also, during mixdownj when tape tracks are no longer monitored but MIXED, the signal path between tape input and main channel is kept to a minimum.

The configuration is 24 into 8 into 24. This means that there are 24 channels, 8 groups or ‘submixes’ (or 4 stereo groups), and 24 tape monitor returns, one for each channel. There are 24 100mrri channel faders, 8 Group faders, and a single stereo fader driving the L R mix.

In remix mode 48 channels are available, all with EQ and access to the aux buses. There are an additional six stereo FX returns, giving a grand total of 60 separate line-level inputs: and that’s before you even consider using the group insert points to provide 8 more !

There are 6 aux buses accessed by four potentiometers, two headphone mixes, and professional recording, monitoring and talkbackl facilities. If you can afford to lose the extra 24 line inputs, the MIX-B bus can also act as a separate stereo aux send, giving 8 aux buses in all.

A comprehensive set of inputs and outputs include MIC (+48 V)j Line, Tape (+4| dBu or -10 dBV)j inserts all round, direct channel outs, and all master recorder and monitoring options. Just about everything you’d expect from a massive console.

In addition, a 1/4" jack expander bay (INPUT ONLY) allows direct patching into all buses within the EURODESK (except SOLO and PFL). Hence two EURODESKs may be linked, or the EURODESK coupled to ANY OTHER CONSOLE, large or small, provided that the other console has (or can be MADE to have) similar access (see Section 19 ‘EXPANDING THE EURODESK’).

Last, and probably least, two BNC connectors atop the console await optional gooseneck desk lights, available from all good gooseneck desk light stores. A must for those darkened auditoria, or when you’re into your third consecutive night in the studio.

2.2 METERING

The 1-24 channels have signal and overload LEDs, while the L and R output and 8 groups each have 12 segment bargraph meters. 0 dB is referenced to the selected tape operating level (+4 dBu or -10 dBV). The MAIN MIX (L/R) meters also have clip LEDs (+28/22 dB: Balanced/Unbalanced), and double up as mono PFL or stereo SOLO meters, or 2T return meters, or EXTERNAL return meters (in general, what you HEAR is what you SEE). During Solo/PFL ONLY the MAIN MIX bargraph meters illuminate.

The master CLIP LEDs (+22 dBu) should never be allowed to illuminate. If they do, reduce EITHER the MAIN MIX fader, or the GROUP(s) and/or CHANNEL fader(s), or (as a last resort) the channel input gain(s). Maybe it’s time to do a round of PFL metering.

In SOLO/PFL mode a 0 dB meter reading matches an internal operating level of 0 dBu (0.775 V). However, when looking at the mix, 0 dB is referenced to +4 dBu, the 2T operating level. I.e. if only ONE signal is present in the MAIN MIX bus, SOLOing that signal will cause the meter reading to increase by +4 dB.

2.3 PSU

Not an infection of the urinary tract, in fact PSU stands for the least glamorous, most frequently underestimated feature of any electronic device: the POWER SUPPLY UNIT. The EURODESK remote PSU connects to the desk at the rear of the console via a multiway connector. 2 1/2 U high, nevertheless it is designed to slot into a 3U rack space. The extra 1/2 U is to allow air to circulate around the heatsinks employed for heat dissipation. If your control room is small, forget heating. The EURODESK PSU chucks out a massive 400 Watts.

Why? Any amplifier circuit is limited in it’s transient response by the available current. In common with most desks of this size, the EURODESK has more than a thousand line level operational amplifiers (op-amps) inside. When being driven hard, many desks begin to show signs of stress due to power supply limitations. Not so with the EURODESK. The sound should stay clean and crisp and TIGHT right up to the operating limits of the op-amps themselves.

Do not connect the PSU to the EURODESK while the PSU is connected to the mains supply.

3. INPUT / OUTPUT CHANNEL

3.1 CHANNEL STRIP

On the EURODESK the 24 INPUT + OUTPUT (I/O or ‘normal’) channels cover most of the console. Most of each strip is occupied by the main or A-CHANNEL, which can accept MIC, LINE, or TAPE inputs, depending on the positions of S1 and S3 (see fig. 3.1a and 3.2). Each channel strip also sports a secondary B-CHANNEL (fig. 3.1e) Anything routed to a ‘B’ channel is directed to a separate MIX-B bus.

(See section 3.7: B-Channel, also section 6.2: MIX-B Master).

3.2 INPUT SWITCHING

Look first at the MIC/LINE switch S1. In the UP position it selects MIC. In the DOWN position LINE. The next switch S3 chooses whether CHANNEL A looks at INPUT or TAPE. If INPUT is routed to Channel A, TAPE is offered to Channel B. If TAPE is routed to Channel A, INPUT is offered to Channel B.

When track laying it is usual to use the MIX-B inputs for monitoring the signal from TAPE, while the A channels take care of microphones, Dl’s etc.. For MIXING purposes, TAPE tracks are normally ‘flipped’ onto the main channels, leaving the B Channels free for other applications, such as offerinq extra line inputs. (These rules are, however, made to be broken.)

S23 applies to the B channel only, and replaces the normal TAPE or INPUT source with a tap from the main channel, taken after Channel Fader and Mute Switch. Now MIX-B acts as an extra stereo POSTFADE AUX send or extra stereo mix. You should remove MIX-B from the Main Mix (via S48, Master Section) in this configuration.

The B channels 25-48 are only routable EN BLOC to the main mix, via S48. Therefore the MIX-B bus can only have one function at any one time, either as a stereo AUX or secondary mix send (S48 UP), or as a set of 24 extra line or tape inputs to the main L R mix (S48 DOWN).

3.3 INPUT GAIN SETTING

The channel input level is set by the TRIMPOT (P2). Use SOLO/PFL (S26) to bring the channel input onto the L + R bargraph meters under the MASTER section of the EURODESK. This also sends the SOLO/PFLed signal to the Left and Right speakers. Channel Solo/PFL (S26) has an associated LED (L26).

(See also 13.1: Channel A setting up procedure and 6.5: Solo/PFL.)

For level-setting (as opposed to localised listening) choose to use the mono PFL rather than the post-fader SOLO bus (S95 DOWN).

SOLO/PFL never interrupts the mix at the main recording outputs. It follows that Aux Sends and Groups must also be unaffected, since they can contribute directly to the Main Mix.

In addition to switchable SOLO/PFL metering, a couple of LEDs (L24 and L25) continuously monitor whether a signal is present (-20 dB), and when the channel is going into overload. These take their cue from three test points: input, post EQ and post fader. In all cases the higher level wins. You do NOT want the overload light to come on, or if it does no more than very intermittently during a take or a mix.

3.4 MAIN EQUALISER

This can be switched (S10) out of circuit for easy A/B comparisons between EQed and straight signals, or for when you know you don’t want to use Desk EQ at all. It is best considered in three sections. First off there are two Baxendall shelving frequency controls for treble and bass, at 12 kHz and 80 Hz respectively (P4 & P9). These are DUPLICATED for the B-CHANNEL (P18 & P19), not merely ‘SPLIT’ off from the main EQ.

I.e. you can have a full 4 band EQ on the main channel AND a two-band EQ on B.

Secondly, there are two semi-parametric swept mids, Q fixed at 1, which cover the bands 300 Hz to 20 kHz and 50 Hz to 3 kHz. An unusually broad frequency range is catered for, and there is an enormous 3-plus octave overlap between the two mid bands (P5, P6, P7, P8). No experienced engineer will complain about that! All 4 bands offer 15 dB of cut and boost.

Thirdly there is a steep High Pass (Lo Cut) filter (S11), slope @ 12 dB/octave, -3 dB @ 100 Hz, for reducing floor rumble, plosives, woolly bottom end etc..

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3.5 AUX SENDS

All 6 Aux sends are mono and post-EQ. They are switchable PRE/POST fader in two banks (S13 & S16). For Aux sends 1 & 2, two dedicated pots (P12 & P13) are used. These can be taken from a point before or after the channel fader, i.e. PRE or POST (S13). Aux sends 3 & 4, and 5 & 6 are serviced by two potentiometers (P14 and P15). The SHIFT (S15) button determines whether buses 3 and 4 or 5 and 6 are addressed. Also, these 4 sends can be derived from the MAIN MIX. or MIX-B, depending on SOURCE (S17), and, as before, can be Pre or Post (S16).

For almost all FX SEND purposes, you will want ‘ Auxes to be POST-fade, so that when a fader level is adjusted, any reverb send from that channel follows the fader. Otherwise, when the fader is pulled down, the reverb from that channel would still be audible. For CUEing purposes, Aux sends will usually be set PRE-fade, i.e. independent of the channel fader (S13/16).

Most reverbs etc. sum internally the left and right ‘ inputs. The very few that don’t may be driven in true stereo either by 1) 2 aux sends or 2) the MIX-B bus (see SECTION 3.7: MIX-B).

There is +15 dB of gain on every AUX send. Such a high boost is only appropriate where the channel fader is set around -15 dB or lower. Here, an almost exclusively WET signal will be heard. Previously, in most consoles, such a wet mix required the use of a PRE setting for the channel Auxiliary Send. This meant losing fader control over the signal.

3.6 ROUTING & MUTING

ROUTING means selecting which BUS you want a channel to address. There are actually 6 stereo buses in the EURODESK (plus a stereo SOLO bus). The MAIN MIX bus is selected by S32 (see figure 3.1d), while the GROUPS are selected by switches S28 (for groups 1 and 2), S29 (3 and 4), S30 (5 and 6) and S31 (7 and 8). Odd and even numbered groups are selected via the main A channel PAN P24, as are the left and right mix buses. (The sixth stereo bus is the MIX-B bus, with it’s own independent pan control P20; see SECTION 3.7: B-CHANNEL). Usually, only one of S28-S31 will be selected for a particular channel. (See fig. 3.3.)

TIP: An exception to this rule is when laying down voice takes. It is often convenient to have the mic channel(s) routed to alt potential TAKE tracks simultaneously, since you are often dropping in quickly between four or more tracks. It means one less button press each time you switch tracks.

Level to the Group and MAIN MIX buses is ultimately determined by the channel faders. These are made by Panasonic®, and are designed to give a smooth logarithmic taper of a type more usually associated with the name of some pretty expensive brand... The low level performance particularly is far smoother than that of a normal 'budget' fader.

The MUTE button (S27), like that for SOLO has an LED indicator, and removes the Channel-A signal from all buses, save any Auxes set to PRE-fader. It is ergonomically placed immediately above the fader, and engaging MUTE is equivalent to setting a fader level of minus infinity

3.7 CHANNEL B

The B-Channel (Fig 3.1e) comprises a secondary channel with it's own hi and lo EQ, pan and level (P18, P19, P20, P21). The EQ is a replica of the A-Channel shelving EQ. The B-Channel ALWAYS feeds into the MIX-B stereo bus, but it's source can be switched between TAPE, LINE, MIC and A-CHANNEL, depending on how 31, S3 AND S23 are set (see Fig. 3.2 and section 3.2). Unusually for an 8-bus console, B-Channels also have their own MUTE buttons (S22). Aux sends 3/4/5/6 may be diverted from the A to the B-Channel via S17. Therefore if the B-Channel is being used to monitor off-tape, some FX processing e.g. reverb and echo can still be applied. (See: 16.3: WET MONITORING.)

When Channel B looks at Channel A (S23 DOWN), the signal comes from after the Channel A Fader and Mute switch. A modification can convert this POST stereo aux send to PRE. (See APPENDIX II: Modification No. 2.)

4. INSERTS

Insert points are useful for adding dynamic processing or equalisation to a channel, group, or the mix. Unlike reverbs etc., which are usually ADDED TO the dry signal, DYNAMIC PROCESSING is normally applied across an ENTIRE SIGNAL. Here an aux send would be inappropriate. Instead the signal is intercepted somewhere along the channel/group/mix, fed through the dynamics processor and/or EQ, then returned to the console AT THE SAME POINT WHERE IT LEFT The insert point is invisible or NORMALISED, until a jack is plugged into it.

All groups and channels have got INSERT points, as does the main stereo output. Both SEND and RETURN are accommodated on a single stereo *M4"* jack socket wired tip=send, ring=return. Inserts are always pre-fade, and also pre-EQ / aux sends for channels.

Insert points may also be used as Pre-EQ direct outputs without interrupting the signal flow. This is obvious when looking at the Patchbay wiring (section 8, fig. 8.1). If you want to insert a dynamics processor POST-EQ, the insert point must either be taken from a group, or via a second channel / aux return as follows:

a) Insert a compressor / gate / EQ across a GROUP, and route the CHANNEL to be processed (and only that channel) to that group.

b) Alternatively, patch a channel's DIRECT OUT into a compressor/gate/EQ. Take the output from that compressor / gate / EQ and feed it back into the desk via a secondary input (channel, aux return etc.).

Figure 4.1 illustrates how you might insert into a channel post-EQ for mixdown, or track-laying (their requirements are different). Mixdown requires one A and one B Channel. Recording requires two A-Channels.

In this arrangement you might find that compression tends to soften the perceived amount of EQ applied. The solution? Apply more EQ. This creates a real 'pressure' sound, great for high energy music such as dance. (For a more subtle approach, use the desk insert points word for word.)

Using a group insert to effect post-EQ processing precludes the use of POSTPROCESSING AUX SENDS without some serious re-patching.

5. GROUP AND DIRECT OUTPUTS

5.1 GROUPS

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The principal routes to the multitrack are via the GROUP OUTPUTS. There are 4 stereo (or eight mono) groups, numbered 1-8. All main channels can access all of them, as can the STEREO AUX RETURNS 1 and 2. (For this reason it is usually wise to bring your best two FX processors back on these returns (or A-Channels, for that matter), so that they can easily be sent to tape.

(See also Section 6.1: AUX MASTERS.)

Q.: Why are there 24 group output jacks on the EURODESK when there are only 8 groups? Well, each group output is triplicated, so that the EURODESK can interface with up to 24 TRACKS via the group outputs without having to re-patch.

As well as ALWAYS functioning as subgroups for track laying via the GROUP OUTPUTS, groups can be ROUTED DIRECTLY INTO THE MAIN MIX BUS for submixing. MAIN MIX routing is handled by switches S37 & S38. S37 routes an ODD numbered group to the LEFT bus, while S38 sends an EVEN numbered group to the RIGHT bus. That's fine for stereo submixes. If instead you want a pair of MONO submixes, press ALSO the MONO buttons (S35 and S36). Now these groups feed into the CENTRE of the L R stereo image, i.e. equally to L and R. You could have the first group feeding into the left hand side, while the second one appeared in mono, but I can't think of many real situations when you'd want to do this.

Group SOLO (S33 + S34) follows the mix assignment. E.g.: if L + R are selected, then that stereo group is monitored IN STEREO, If MONO is also selected, monitoring is in MONO.

TIP: Try inserting compression / de-essing / an exciter / a gate across grouped signals (e.g. backing vocals, drums, layered synths).

TIP: Try merging a dry signal with a little wet, then compressing the sum heavily. Though the reverb proportion will be low when a signal is present, the resultant reverb tail pumped up by the compressor at the start of each SILENCE will give the illusion that the reverb was massive alt the time. (The listener will be left wondering how the singer could sound so clear in such a wet acoustic!)

EXPERT TIP: EURODESK insert points are, of course, simultaneously inputs and outputs. I've NEVER understood why mixer manufacturers insist on putting these onto a SINGLE STEREO JACK SOCKET. I suppose it saves a little money, but I suspect it's more BECAUSE THAT'S THE WAY IT HAS ALWAYS BEEN DONE. For goodness sake, get them onto a PATCHBAY, where they can appear as independent sockets, and do away with all these fiddly Y-leads that always seem to be the first to get knotted in the flightcase. (See Section 8:PATCHFIELD.) Now it is possible to do the following incredibly useful patch without having to make up what would amount to a ring-to-tip, tip-to-hng stereo patch lead.

5.2 DIRECT OUTPUT

Each of the 24 main channels on the EURODESK has its own DIRECT output, which is taken from a point immediately after the fader (i.e. post EQ and after the aux sends, see fig. 3.3). This can feed a tape track directly without having to resort to the groups, enabling more than 8 different tracks to be recorded simultaneously. Almost alone among the EUROjacks, these are on UNBALANCED MONO sockets at +4 dB. (See APPENDIX I, Table No. 2, also Section 16.1 'Recording').

6. MASTER PANEL

6.1 AUX MASTERS

6.1.1 AUX SENDS

Much of the Master section (situated above the bargraph meters) is taken up by master aux sends and returns. We'll start with the sends (see fig. 6.1.1).

Stacked in a vertical column are six master aux send levels, one for each of buses 1-6 (P41-P46). Each has a gain structure of -infinity to +15 dB. The extra 15 dB of gain comes in once a knob passes a centre detente (representing the 'normal' unity gain position), enabling insensitive outboard FX to be properly driven. Each AUX send has a SOLO button (S41-S46), and, as with other areas of mixer, a LOCAL SOLO LIGHT (L47), which illuminates when any of the AUX master sends are solo-ed.

(This is to help you see exactly WHAT has been solo-ed. Any experienced engineer will have had occasion to search painstakingly through EVERY solo button on his / her console trying to find out why the main solo light was on, and the control room monitors silent!)

6.1.2 AUX RETURNS

Across from the AUX sends are the STEREO AUX RETURNS (see fig. 6.1.2). These can be thought of as a dozen extra line inputs configured as six stereo pairs. On these inputs there is up to 20 dB of gain available. Alternatively, a MONO (centre-panned) signal may be returned by plugging into the LEFT Aux return jack only.

This feature is disabled if all tine-level t/Os from the EURODESK are wired permanently to a PATCHBAY (see SECTION 8).

a) 1 & 2

Aux Returns 1 & 2 have full group routing matrices to enable returning FX to be sent to tape, plus L R bus assignment. The functions for AUX Return 1 (mirrored by AUX Return 2) are: ROUTING (S49/50/51/52/53) LEVEL (P49), BALANCE (P51) and

SOLO/PFL (S54). LEVEL controls the amount of signal being blended into the mix or a group, while BALANCE controls the relative amounts of L and R processed signal. I doubt if you'll often want to adjust BALANCE away from centre.

EXPERT TIP 1: As always, there are exceptions to the above rule. Some short stereo delay effects (say 30 ms to L, 50 ms to right) cause a psychoacoustic effect where the EARLIER delay seems louder. A similar effect is noticeable when harmonising in stereo: a slight pitch shift upwards will seem louder than one that goes down. In both cases use the BALANCE *control (P51) to compensate. (An analogy comes from Greece: the columns of the Parthenon in Athens are slightly bowed so as to APPEAR straight.)*

EXPERT TIP 2: When carrying out expert tip No. 1, or any other stereo imaging exercise, don't just rely on the control room monitors. Get a pair of headphones and listen in stereo AND in reverse stereo, to allow for any hearing discrepancy between your ears.

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b)3&4

And so to AUX RETURNS 3 and 4. These too have a routing matrix (S55-57 for AUX Return 3), but this time it is designed to facilitate MONITORING rather than RECORDING. The options are MAIN (L/ R) MIX, and headphone mixes 1 and 2. Gain pots and solo switches complete the picture.

c)5&6

AUX Returns 5 and 6 are the poor relations, with only a level pot and solo button each. These are always assigned to the MAIN (L/R) bus.

d) SOLO

Below each column of AUX returns lies a local solo LED (L61 & L74). These illuminate whenever a SOLO button in the column above is pressed.

There is no absolute reason why the send from AUX 1 should feed into a processor whose outputs are sent to the AUX 1 RETURN. The processor could just as easily be patched into the AUX 3 Return, or even a pair of channels. For many purposes, however, it is sensible to set up a default patch where the AUX outputs and inputs correspond. It is logical to put your premier FX units into the Aux 1 and Aux 2 loops, since these returns enable you to record to tape without re-patching.

TIP: An exception to the above is when recording a group of performers live to multitrack. (See Applications SECTION 16.3: WET MONITORING).

TIP: Sometimes an engineer wants to narrow the stereo width of a reverb field. To do this you will have to come back on either A or B Channels, which have full PAN facilities.

6.2 MIX-B MASTER

Only 2 controls occupy the MIX-B Master (fig. 6.2). P48 offers the standard EURODESK gain of up to +15 dB. S48 is crucial: it routes the MIX-B bus output into the MAIN (L/R) bus. MIX-B can have three basic functions:

a) It can act as an entirely separate mixer-within-a-mixer to provide a completely separate mix (S48 UP, S23 DOWN). (See also SECTION 17.)

b) It can act as an additional stereo Aux feed to FX during mixdown (S48 UP, S23 DOWN).

c) It can provide 24 extra B inputs to the mix (S48 DOWN, S23 UP).

TIP: For live applications try using MIX- B to feed a secondary set of speakers. These could be sidefills, or more spectacularly, the rearward portion of a quadraphonic sound system.

There is no SOLO provision for MIX-B. However, you can AUDITION it by selecting only MIX-B (S83) in the MONITORING sourcing matrix.

If MIX-B is assigned to the L H mix (S48 DOWN), DO NOT listen to MIX-B (S83) AND the L R mix (S82) simultaneously. That way you'll be monitoring MIX-B twice over, and what you hear WON'T correspond with what's going down to tape.

6.3 MONITORING

Though most of you will want to audition the L R mix most of the time there are exceptions. These include SOLO/PFL, and 2-TRACK PLAYBACK. The SOURCING matrix (S82-S85, see fig. 6.3) allows you to monitor the main mix, the B mix, and two external sources marked 2-Track and EXTERNAL. The L R meters follow whatever source is being auditioned. (The meters won't make much sense if more than one source is selected!)

TIP: EXTERNAL could be 'normalled' to a hi-fi pre-amp, allowing you to monitor extra sources such as vinyl, cassette, CD etc..

Altering what goes into the control room's MONITORS does not affect the signal from the L R recording outputs. Just as well, or every time you wanted to do a quick SOLO during a mix, you'd have to start again!

A ROOM monitor volume pot P86 sets the level to the Control Room monitors. This is sourced POST the main MAIN MIX stereo fader setting: otherwise you wouldn't be able to hear your fades. There is also a similar STUDIO volume pot (P82).

TIP: Owners of MIDI production suites might like to drive a second pair of Control Room speakers from the Studio output, but take care when using the TALKBACK mic: no -20 dB offset is applied to the Studio output!

Me: I'd have half-a-dozen sets of speakers on an external switching matrix, including NS10M, Ghettoblaster, Club system, Car Stereo and overblown *2"* speaker loosely screwed into a less-than-airtight cardboard box.

If you are using the STUDIO output to drive a pair of monitors actually IN the studio, DO NOT EVER leave P82 turned up during a take. Howls and howlround may well be the result.

Lastly, there is a MONO button (S86), useful for checking the phase correlation and/or coherence of a stereo signal. Again, this does not affect the MAIN MIX output.

6.4 HEADPHONES

Both HEADPHONES 1 & 2 Masters are identical, i'll describe the first.

A SOURCING matrix picks up any or ail of MIX-B (S76), CONTROL ROOM (as chosen in Monitor section, S77), AUX 3/4 (S78), AUX 5/6 (S79) and EXTERNAL (S80). In addition to the sources which are directly selectable from the Headphones Masters, AUX Returns 3 and 4 may be 'force-fed' into HP1 & HP2 from the Aux Returns Masters (S55/56/68/69).

The headphone mix level is controlled by a master volume pot (P75), and the gain is sufficient to drive headphones directly. This is fine for a MIDI suite with overdub booth, but for the bigger studio's headphone network I'd recommend using a separate headphones distribution amplifier like the Behringer HA-4000. This can offer the added advantage of independent headphones level control for every performer.

A SOLO button (S81), with its own LED, enables monitoring of the headphones amplifier's output signal.

This way the engineer can monitor what's going on in the cans on the control room monitors, though in my experience this does not give as true a picture as auditioning the cue feed from a set of headphones identical to those worn by the performer(s).

6.5 SOLO/ PFL

6.5.1 SOLO

SOLO is short for SOLO IN PLACE, and is the preferred method for auditioning an isolated signal, or group of signals. Whenever a SOLO button is pressed, all unselected channels are muted in the monitors. Stereo panning is maintained. The SOLO bus is derived from the output of the Channel PANs, AUX Send/Return pots etc. etc., and is always POST-FADE.

E

6.5.2 PFL

Pressing S95 disengages the stereo SOLO bus, and replaces it with a separate mono PFL (Pre-Fade Listen) bus. Now ANYTHING AT ALL which is SOLOed, isn't. It is PFLed instead. PFL should be used for gain-setting. (See also the essential Section 13, START-UP.)

In addition to any LOCAL Solo LEDs which might be activated, the ultra bright MAIN SOLO LED (L95) illuminates whenever ANYTHING is SOLO/PFLed.

P94 controls the master SOLO/PFL level. Set to unity gain (centre detente), this will match the mix level.

6.6 TALKBACK

The built in mic (above the Main Mix Fader) allows you to converse with artists remotely. Most important controls are the VOLUME (P99) and PHONES & STUDIO button (S99, see fig. 6.6). It is possible to route the talkback mic to any of the following: Aux 1, Aux 2, Groups, and Phones & Studio (S96-S99).

Complex headphone or stage monitoring networks could be constructed where HP1, HP2, Aux1 (PRE), Aux2 (PRE) AND one or more GROUPS all fed separate monitor mixes. (See Section 16.2: VERY TRICKY HEADPHONES). However, since the 4 pushbuttons are NON-LATCHING it might drive you crazy. If you ARE using many headphone

feeds, it's possible you will find it easier to patch a separate microphone for talkback straight into a channel, where it can be routed pretty much anywhere. I'd have an SM58 handy, myself. Minimise feedback by placing the mic 0.001 mm away from your mouth,

Engaging TALKBACK (S99) dims the control room (monitors, not lights you fool) by -20 dB to restrict the possibility of howlround. All other Talkback routes are unaffected.

7. CONNECTIONS

7.1 REAR PANEL LEFT

PSU

The chunky remote 19" racked power supply is connected via a circular multiway at the left-rear of the console. This supplies several DC operating voltages incl. +/-18 V (audio circuits), +48 V (mic phantom power), +12 V (BNCs) and +5 V (LEDs).

THIS CONNECTION FOR BEHRINGER

POWER SUPPLY ONLY !

Expander port

Access is given to all subgroup, aux and mix buses via an array of 1/4" jacks, situated on the back panel.

Main Balanced Outputs

XLR, for connecting to the mastering recorder. Wired pin 1 ground, pin 2 hot, pin 3 cold. Maximum level is +28 dBu. Rear panel mounted.

7.2 REAR PANEL RIGHT

Tape sends / Group outputs

The 8 subgroups are each connected to THREE stereo jack sockets, for easy patching into 16 or 24 track recording systems. Once again operating level and configuration is switchable, in two banks of (3 x) 4.

Inputs B / Tape returns

These also accept balanced or unbalanced 1/4" jacks, and are switchable, in groups of 8, between -10 dBV (unbalanced) and +4 dBu (balanced), corresponding to the standard semi-professional and professional operating levels and configurations respectively. Consult your multitrack manual to find out which one applies. The switches are to be found immediately after each array of 8 tape / line sockets on the back panel.

7.3 TOP PANEL LEFT

Mic inputs

These are via XLR-type connectors, wired pin 1 earth, 2 hot, 3 cold, for BALANCED LOW-LEVEL operation. Since most quality capacitor mics require a 48 V DC offset to charge the plates, PHANTOM POWER is provided. This can be switched on or off in 3 blocks of 8 via a switch above the XLR for channels 1, 9 and 17.

Care should be taken NOT to plug mics into the console (or stagebox) while the phantom power is on. Also, mute the monitor I PA speakers when turning phantom power on or off. Allow 1 minute after powering up for the system to equilibrate before setting input gains. The mic input is situated above the channel strip.

Channel A line inputs

These accept balanced or unbalanced 1/4" jacks, tip = hot, ring = cold, sleeve. Situated above the channel strips.

Channel-A Direct Out

This tap comes from just after the channel fader. Unbalanced 1/4" jack.

Channel Inserts

These provide for unbalanced send and return from a single stereo jack socket. Wiring is: tip = out, ring = in, sleeve. Situated above the channel strip.

BNC lamp connectors

For optional gooseneck desk lights (e.g. Littlelite® lamps # 12G or 12G-HI). The lamp sockets are wired centre post +12 V.

7.4 TOP PANEL RIGHT

Group Inserts

As for Channel Inserts, but for Subgroups. Situated above the Master Section, top row.

Main Inserts

As above, but for the MAIN (L/R) MIX. Situated above the master section.

Auxiliary sends

Unbalanced 1/4" jacks, situated immediately below the group inserts. They operate at+4dBu.

Auxiliary returns

All six stereo pairs are on unbalanced 1/4" jacks, and operate at +4 dBu. Situated immediately below the aux sends.

2-Track and External inputs

Unbalanced jacks. Bottom left of master patch, above master section.

Stereo outputs

Control Room Output (Monitors), Main Mix, Mix-B and Studio Output. All are on a pair of unbalanced jacks, nominal put level +4dBu. The main mix is also available as a fully balanced pair (see 7.1).

Stereo headphone outputs

These are on stereo jacks, wired tip = L, ring = R, sleeve = ground (see section 6.4:

HEADPHONES).

7.5 PLUG SOLDERING GUIDE

You will need a lot of cables for a lot of purposes. Here's how they should look:

You'll need this kind for:

Expander Port Bus Inputs

Subgroup Outputs I Tape

Sends

Direct Out

Aux Sends

Control Room Out

Main Mix Out

Mix-B Output

Studio Output

Aux Returns

2-Track Input

External Input

W stereo (TRS) plugs

You'll need this kind for:

Inserts

Subgroup Inserts

Main Inserts

You'll need this kind for:

Headphones

You'll need this kind for:

Line In Inputs-B
Tape Returns

Maybe you will need this 'Y' shaped thing for Insert purposes

To mixer channel *insert*

To

processor

input

From

processor

output

Female XLR plugs (‘soldering’ view of the cable connector)

1= Ground/bcreet 2= Hot (+ve) 3= Cold (-ve)

1= Ground/Screen 2= Hot (+ve) 3= Cold (-ve)

You’ll need this kind for Main balanced outputs

Male XLR plugs (‘soldering’ view of the cable connector)

1= Ground/Screer 2= Hot (+ve) 3= Cold (-ve)

You’ll need this kind for Mic input

Be sure to read sections 8 and 12 BEFORE you start heating your soldering iron!

8. THE PATCHFIELD

Nomenclature clarification:

FIELD = entire patching area BAY = a unit of 48 jack sockets arranged as 24 outs over 24 ins.

If you really want to make the most of your home studio, invest in a patchfield. I know t am often put off doing a complex patch if there is no patchfield:

1) because it’s so much bother! and

2) in case I inadvertently damage or pull out a lead.

8.1 THE NORMALISED BAY

Most decent Jackbays offer two rows of 24 NORMALISED jacks in a 1 U of rack space. Lucky you’ve got a 24/48 channel desk, eh? The term NORMALISED refers to the fact that the TOP row (Outputs) are internally connected to the BOTTOM row (Inputs), UNLESS you plug something into an INPUT socket. Plugging into the

OUTPUT socket of a normalised insert pair does not break the internal connection, but it does provide an alternative DIRECT OUTPUT Where normalising is NOT wanted on a Patchbay (there are a few cases!) it is possible to remove it by cutting certain PCB tracks. Refer to the Patchbay instructions for how to do this.

8.2 THE PATCHFIELD

If you want to do the only decent thing, and construct a Patchfield for your studio, here's how to do it. Note that I have laid it out in order that a minimum number of cables are likely to be needed. I have also completely left out the microphone inputs. Unlike everything else, these operate at a level several orders of magnitude lower than line (+4 or -10 dB). It is best to plug mics directly into the EURODESK, or via special XLR-type wal! boxes connected to the EURODESK mic inputs by a good quality balanced (2-core + screen) multicore. (See also Section 12: (UN)BALANCED LINES.)

*

Break the normalising on this bay. **

Dri-\'-il/ thf-\ n/-i r rin'-i 11

Tie lines: usually, in a MIDI setup, racks and keyboards etc. are scattered around the control room. Plugging these directly into the front of the patchbay would result in Spaghetti Junction. Instead, better to connect TIE LINE jacks to wall boxes strate-gically positioned near to where MIDI hardware congregates.

if * *

It's always good to have a few 4-way links around for splitting signals up to 3 ways (one in, three out). E.g. one tape track has four different instruments on it. Patch the DIRECT out of it's CHANNEL into a 4-way split, returning to a further 3 channels via LINE INPUT. Set up each of the paralleled 4 Channels for one instrument, and use mutes (preferably MIDI controlled) to mute the 3 unwanted channels at any one time).

| *Break the NORMALISING LINKS on positions 21 thru 24. Note also that the MIX B* I *outputs are adjacent to the AUX sends. This is because one of the two functions of MIX-B (source switches set to CHANNEL) is to provide an extra stereo aux send.*

TIP: You can treat MIX B as two mono sends using GAIN for level and PAN for blending. Setting pan to the centre will give a 50/50 ratio of the 'Aux 7' and 'Aux 8' effect, hard left 100% 'Aux 7', etc..

icicle*

Break NORMALISING LINKS here. If you've got more than 5 or 6 stereo dynamics / EQ processors, you might spill over onto another dedicated bay, or alternatively have to find some suitable extra space somewhere else in the patchfield. Remember, most dynamics processors also have sidechain/KEY inputs, and therefore require 3 holes per channel.

TIP: Enhancers are usually applied across INSERTS like compression and EQ etc., but most BEHRINGER Exhancers have a SOLO mode, in which they can be addressed via an AUX send and blended back into the main mix like any other REVERB.

Bay 8: A lot of clever stuff is going down here. I'd better explain:

L7 & R8 equals the L R REC INPUT to the 2-TRACKS. On Bay 8 I have HARD-WIRED these to REC OUTPUTS 1-6 in order to drive all recorders simultaneously. Copying from ANY 2T source to ALL recorders may be done by patching the SOURCE outputs into L7 & R8.

I have assumed you have a HIFl amp available to enable a variety of secondary sources to be condensed into the XTRNL (External) input for easy monitoring selection via the HIFl amp's input selector switch, if you want to RECORD from any of these sources, best patch direct from the individual outputs (17 - 22) rather than the HIFl amp mix (15 & 16) for the cleanest result. (The exception being VINYL, which will need to use the HIFl amp's RIAA Pre-amp to present the mixer with a flat response signal).

I'll bet you never realised how BIG a patchfield you really needed!

8.3 LOOMING PROBLEMS

Loom wiring is an art unto itself, and it is worth taking time out to get it right. First off, it is important to avoid earth loops. (A looped wire acts as an ariel, picking up electromagnetic radiation.) Think of a tree. Every part of that tree is connected to every other part, but only by one route. That's how the total earth picture for your entire studio should look. Don't take the earth off your power cable plug to reduce audible 50 Hz mains hum (or it's harmonics). Rather you should be looking at disconnecting the SIGNAL SCREEN somewhere.

You could do worse than ensure that ALL screens are commoned at the patchbay in which case all unearthed equipment would pick up earth from this point via a SINGLE screen (more than 1 route == an earth loop), while mains-earthed equipment would have all screens CUT at the EQUIPMENT end.

Some quality equipment has an independent signal and mains earth. In this case at least one, and possibly all, screens should carry earth to the equipment. Sometimes the only way to find out is 'suck and see'.

Take care to ensure that using the patchbay does not disturb the studio's earth architecture. Always use SHORT AS POSSIBLE patch leads with the screen connected at both ends.

If you're really serious about hum levels, you could run BALANCED lines wherever appropriate. The earth wiring scheme would be the same as before. By shorting the ring to the barrel for all balanced jack SOCKETS connected to unbalanced EQUIPMENT, you could use BALANCED patch leads throughout. (There is no percentage in wiring a balanced output to a balanced input with a MONO patch cord!) (See Section 12: (UN)BALANCED LINES)

Having designed mains hum out of the system, make up your cable looms from the patchbays outwards, and use cable ties, flexible sheaths, multicores etc. to keep the back of your racks tidy. It's going to get VERY busy in there, and loose cables will inevitably mean lost signals. Possibly even lost equipment!

9. EQUALISATION

The variable parameters of the channel A and B equalisers on the EURODESK are described in Sections 3.4 and 3.7.

Few people buying the EURODESK will need to be told how an equaliser works. But how to get the best out of it? Weil, that's another story.

In the beginning EQ was an instrument for removing unwanted frequencies, or compensating for imperfect microphone response curves, or bumps in a studio's acoustic. It was a corrective device. Tamla Motown turned that notion upside down in the sixties with the novel idea that you try to find for each instrument a characteristic frequency not shared by the other instruments in the mix. Then you whack up it's gain. This makes individual voices punch through a mix in a slightly unnatural but exciting way. In general corrective EQ usually involves broadband (slope) contouring, together with narrowband notching of unwanted resonances. The narrower the notch or 'Q', the less the total signal will be affected.

TIP: Finding bad resonances is made easier by first frequency sweeping in BOOST mode.

'Motown' EQ is achieved by applying boost in a fairly broadband way. The broader the band, the more musical but less instrument-specific the effect. Applying boost over a narrow bandwidth will sound 'Honky'. The two semi-parametric bands of the EURODESK EQ have had their Q fixed at1, a typical and sensible value. For sounds which require drastic corrective EQ (remember no MIDI instrument should need it), it is advisable to have a couple of channels of fully comprehensive equalisation in your rack. (You can always bounce tracks though the outboard EQ, freeing up the unit for the next task). Check out the BehringerULTRACURVE, which promises to be another price/performance buster.

The EURODESK EQ might be applied to a signal as follows: First, trim the LF and HF shelves to achieve the required SLOPE or 'LOUDNESS'. (These controls mirror the tone controls of a typical HIFI amp.) Now use one parametric band to boost the NICEST frequency, and another to cut the NASTIEST. Over all channels in the mix, if too many of the NICEST frequencies coincide, then you might have to settle for second best in some cases! Often you might want to boost TWO nice frequencies. REALLY nasty frequencies will need notching. Time to go outboard.

Q?: Why does the upper mid bell frequency go up to 20 kHz? A more pertinent question might be, why has it taken so long to get there? After all, even 16 bit (the lowest acceptable quality) digital audio sports a 20k bandwidth: surely if 20k is important then SO IS CONTROLLING IT. OK. You and I will never hear a pure 20k sine tone. However, Rupert Neve, the audiophile Guru, would argue that when it comes to real *instruments, what happens even above 20k may have a perceptible effect on the listener. It seems that one reason why high- quality (1/2" at 30 IPS) analogue tape sounds better than DAT to many discerning ears is because, although it's frequency response begins to roll off at 12 dB/octave somewhere around 15-20 kHz, it is not abruptly cut off at 20!*

I have heard, or 'detected', a 20+ kHz low pass filter being switched in and out when monitoring an analogue master tape through a speaker system that included Piezo-Electric tweeter elements capable of reproducing up to 40 kHz. Perhaps less controversially it can be shown that if cut/boost is applied at 20 kHz, a significant portion of the resulting EQ curve for all but the tightest of Q's actually occurs in the audible spectrum, below 16-18 kHz. For example if the Behringer EQ is boosted by +15 dB at 20 kHz, the amount of boost at 10 kHz will be 3 dB. The resulting EQ curve will bear no relation to one where 3 dB of boost is applied at 10 kHz.

TIP: A good vocal signal can be enhanced by applying a significant boost in the 15 k region or higher, ABOVE the nasty sibilance region. ESPECIALLY effective if you've got a de-esser post-EQ.

TIP: Use the LF cut to tighten up channels in a mix: maybe remove it only for the bass, kick drum, toms, tablas, didgeridu and other deliberate subsonics. (When recording classical music ignore this advice).

TIP: With an LFcut at 100 Hz and a LO SHELF BOOST at 80-160 Hz, you have pretty much got a PEAK response rather than SHELVING at the bottom.

TIP: Look at the extraordinary width of the frequency sweep of the upper mid EQ -300 Hz all the way up. Set to maximum boost and play about with the frequency in real time. I bet you'll get some stunning filter sweeps. Try it on drumloops - great for dance music!

TIP: You can cascade channel EQs by connecting the DIRECT OUT (see Section 7:

CONNECTIONS) of one channel into the LINE or TAPE INPUT of another. The first channel should first be un-routed to all buses, including L R and AUX sends. The second channel then becomes the 'control' channel, routing to the buses. You now have a 23 channel mixer, but one channel has a 4 band (semi) parametric plus 30 dB of shelving swing!

Remember EQ contouring can be done with CUT as well as BOOST. E.g.; cutting away the top and bottom, then pushing up the gain is equivalent to MID-RANGE BOOST! EQ is NOT a 1-way street!

Always re-set a channel's input gain after altering the amount of EQ cut or boost applied (see 3.3).

10. GAIN OPTIMISATION

PFL (Pre-Fader Listening) is THE way to set a desk level. Setting up the channel input gain is discussed in the essential Section 13. Optimum master aux send levels will be dependent on the sensitivity of the FX device being driven, but unity gain is a useful starting point. As the mix progresses, more and more channels are likely to be sending to effects via the AUX buses, and it's best to PFL all sends (Aux, Group, MIX-B) just before setting up for the final mix.

Outboard reverbs etc. should all be made to work hard. There's no point in having an 85 dB dynamic range if the input meter of your reverb is barely flickering. On the other hand, digital distortion is not one of the nicer noises around. Fortunately you can SOLO the FX returns. Here you'll have to rely on your ears to detect digital distortion, since different outboard processors calibrate their meters differently, and their dynamic range is not sufficient to allow, say, 15 dB of headroom (as is the case with DAT etc.). The SOLO/PFL meter, on the other hand, looks only at the desk's analogue AUX INPUT level, if you hear distortion, but the meter says you're just hitting 0 dB, then it must be coming from the AUX send amp or the FX unit. If PFL on the AUX send reveals nothing amiss, turn down the input on the FX unit, and turn up the desk's AUX return.

TiP: 99 times out of 100 distortion in the aux send > fx > aux return loop will come from the fx unit (FX GAIN TOO HIGH), and the same goes for a high noise level (FX GAIN TOO LOW).

TIP: Noisy FX (or synth) returns can be greatly improved by the addition of single-ended noise reduction between FX output and aux (or channel) returns. The Behringer DENOISERS are ideally suited for this purpose.

TIP: I've found that using analogue single ended noise reduction can help warm the sound of certain digital reverbs which sound too cold/metallic, and also give that 'Echoplex' sound to digital delay decays.

TIP: Analogue multitrack tape should be driven quite hard, since it's dynamic range (without noise reduction) is likely to be 20-30 dB worse than other elements in the recording chain. Try to record bright. You can always mix back duller. Brightening up an off-tape signal will bring up the level of tape noise.

When mixing or recording, keep the channel FADER levels around or below 0 dB. If you do find the faders creeping up or down, apply a suitable offset over all channel faders, and try to control your bad habit in future !

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11. IMPEDANCES AND TUNING

Electronic Inputs tend to have impedances measured in tens of kiloOhms. Outputs, on the other hand, are generally two or three orders of magnitude less. This is just as well, otherwise a signal at an output might find that the line of least resistance is the limit of the preceding unit.

In the Patchbay section I recommended that you parallel the MAIN MIX output of the EURODESK into alt 2T recording inputs. It would not do any harm to buffer each output from the primary one (i.e. that feeding into your most expensive DAT recorder or 1/2" mastering machine) with a 470 Ohm resistor. Cassette, DAT and Reel to Reel recorders' input impedances should be similar, but just in case they aren't, better to add a fraction of a dB of thermal noise to the inputs of the secondary recorders in the shape of a resistor, rather than having an unusually low impedance input grabbing most of the signal. Another neat idea is to parallel the Monitor L R output via a 47 kOhm resistor pair. Now you can safely connect e.g. a tuner to either extra hole, without shifting the stereo image (this would happen if a low impedance tuner input was connected directly across one side of the monitor output). Now, whenever you monitor an instrument's input level with the SOLO/PFL function, you can check it's TUNING also. That should impress the customers. Especially those using old, unstable, but very desirable analogue synths.

12. (UN)BALANCED LINES

Balanced inputs and outputs are offered on most audio connections on the EURODESK (Inserts and Direct Outs being the major exceptions).

Why? Though all audio cables (except speaker cables) have earthed screens, the shielding they afford from the electromagnetic garbage that permeates the atmosphere is never perfect. The balanced line is a simple but effective mechanism to overcome this problem. Instead of one insulated audio conductor, two, usually twisted together, are contained within a single screen. One conductor, wired to pin 2 of an XLR/Cannon connector by international convention (after decades of total confusion!) carries a signal variously referred to as 'hot' or 'positive'. Pin 3 is wired to the 'cold' or 'negative' conductor.

What does this mean? Consider an unbalanced line. Now, that's much easier to understand. You have one 'hot' or 'positive' core, and an earthed screen. The 'hot' wire's waveform, if looked at on an oscilloscope, would be directly correlated to the AUDIO SIGNAL waveform. If you looked closely at the trace, you 'd see random noise along the X axis. What you probably WOULDN'T see, however, is any superposition of 50, 100 Hz etc. corresponding to mains hum interference, since these frequencies would be tangled up in the audio signal (to spot them VISUALLY you'd need to perform a FOURIER TRANSFORM). AC mains frequency and its overtones are picked up by any wire, and some will always leak through a cable screen. The question is, when does it become audible?

Well, all other things being equal, the amount of mains hum picked up by a cable is INDEPENDENT OF THE SIGNAL LEVEL. Speaker lines run 50 or more volts, enough to diminish the effect of mains radiation to vanishingly small EVEN WITH NO SCREEN. (In fact, at these voltages another effect comes into play: capacitive resistance. It is positively undesirable to use screened cable to wire an amp to a speaker. Speaker leads should be as THICK and SHORT as possible, with XLR or WOUND POST terminals.) Line level signals can usually be run unbalanced over short or moderate distances (rack to desk etc.), but NOT from the back of the hall to the stage, ALWAYS PROVIDED that there are no EARTH LOOPS (see Section 8:

LOOMING PROBLEMS: A loop acts as an ariel, positively inviting electromagnetic radiation to flow around the system). MICROPHONE LINES, however, are another story altogether.

Most mics generate not volts, but millivolts. Protecting such a low level signal requires a more sophisticated solution. Hence all mic networks run along BALANCED lines. It works like this. The mic diaphragm moves forwards and backwards according to the air pressure increases and decreases that constitute SOUND

WAVES. Diaphragm movement generates a corresponding electrical signal, which is either POSITIVE or NEGATIVE depending on the direction of travel. The +ve and -ve signals are MIRROR IMAGES of each other: if you shorted + and - you'd end up with nothing:

one would cancel out the other. In fact this cancelling effect is what makes the balanced line work. Instead of simply shorting the negative line to earth, as would be the case in an unbalanced system (losing half the signal, or 6 dB, in the process), the two lines are kept apart until they reach an electronic (or transformer) balanced input.

Here something exquisitely simple happens:

You may not know this, but whenever a signal is amplified, its polarity is reversed. By inverting the NEGATIVE side and adding it 1:1 to an unchanged POSITIVE, a balanced input wastes none of the available signal energy In doing so, it also SUBTRACTS all the radiation picked up along the line. Random noise is unaffected, but you'll hear no hum, and much reduced thyristor noise (from poorly-screened lighting dimmers). Live, you could not run a rig without balanced mic lines, and although in the studio cable runs are shorter, the recorded medium's demand on signal to noise is far greater.

When patching a BALANCED input/output to an UNBALANCED one, simply short the -ve and screen together at the unbalanced input or output.

13.START-UP

13.1 CHANNEL A SETTING UP PROCEDURE

13.1.1SELECTING INPUTS

MICROPHONE: MIC/LINE switch (S1) UP, FLIP switch (S3) UP LINE: MIC/LINE switch DOWN, FLIP switch UP TAPE: FLIP switch DOWN

13.1.2INITIALISING CHANNEL FOR GAIN-SETTING

1) Set gain (P2) and all aux sends (P12,13,14,15) to OFF (fully counterclockwise)

2)EQswitch(S10)UP(off)

3) LOW CUT switch (S11) ON for mics, OFF for signals with desired low frequency content

4) SOLO MASTER set to PFL with PFL bus switch (S95 DOWN)

5) SOLO (S26) switch UP (Light off)

6) CHECK THAT MAIN SOLO/PFL LAMP (L95) IS NOT LIT

7) CHANNEL SOLO switch DOWN (L26 & L95 should light)

13.1.3AUDITIONING A SIGNAL

1) Make a typical noise, or roll the tape. The -20 dB light should flicker, showing that a signal is present. There should also be some activity at the MAIN MIX bargraph meters, indicating the SOLOed level.

2) For MIC/LINE INPUTS: Adjust the GAIN control (P2) until transient peaks are regularly hitting 0 dB.

3) TAPE inputs do not pass through the gain pot (P2). This is why it is important to match the operating level of the desk (-10 or +4) to that of your machine. If the signal is low (due to incorrect operating level setting or too low a level having been recorded to tape), try the -10 setting. If too high, try +4. If neither gives a good level, try patching the TAPE TRACK OUTPUT into a LINE INPUT, and repeat steps 13.1.1& 13.1.2.

4)IfEQisused, repeat steps 13.1.1 & 13.1.2.

5) If an INSERT is used to patch in a compressor, gate, EQ etc, use any outboard processor's BYPASS or EFFECT OFF switch to A/B monitor the effect. Adjust the processor's output level so that effected and bypassed signals are level matched.

6) SOLO switch (S26) UP. Move onto next channel.

13.2 DESK/ TAPE SETTING UP PROCEDURES

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13.2.1 DESK NORMALISATION

All board settings should be set to the normal default condition before or after every session. Usually Faders are set to zero (minus infinity), EQ's set flat and switched out, trimpots and channel aux sends turned fully anticlockwise etc.. The natural initial setting for some pots, EG Master AUX sends, is unity gain. However, some settings, such as selecting PRE or POST for channel aux sends and whether TAPE or MIC/LINE is flipped onto channel B etc. will depend on the operating environment, whether in a MIDI or A/V suite, 24-track studio or live venue, or even just on the engineer's preferred way of working. Ultimately the object of the excercise is:

13.2.2 MULTITRACK INITIALISATION

Set up the multitrack so that any track in 'record ready' condition has its input monitored when the tape is stationary. Place all tracks to be recorded into 'record ready' status (once a recording has been made, these tracks should automatically switch to tape playback). Check that the input levels to each TRACK are optimised before recording commences.

13.2.3 RECORDING LEVELS

When recording to digital, it's a good idea to keep the recorder's PEAK READING meters BELOW 0 dB. Engage 'peak hold' on your recorder if you want to confirm that you haven't inadvertently overstepped the mark during a take or mix. Most digital recorders (though not SAMPLERS) read 0 dB with some headroom left. This is because, unlike with analogue, the onset of digital distortion is as sudden as it is horrible, and the manufacturers of digital recorders don't want you to hear this! If you REALLY want to take your recording level to the limit (and fully exploit digital's 96 dB dynamic range), you'll have some calibrating to do. You could run a tone at 0 dB from the mixer, and use that as your DAT or ADAT reference. But your DAT or ADAT may still be 10-20 dB off it's headroom limit. Probably a better way to work out just how hard you can drive your recorder is to incrementally increase the record level UNTIL IT BLOODY WELL DISTORTS, subtract, say 6 dB, and NEVER EVER EXCEED THAT LEVEL.

When recording to analog, the tape machine's VU meters should show around +3dB on BASS, but only around -10 dB for HI HAT. Although ANALOGUE distortion is more like compression at modest overload levels (often desirable on bottom end), higher frequencies cause saturation even at modest levels (an unpleasant 'crunchiness'). Also, VU meters tend to progressively under-read above 1 kHz, due to their sluggish response time.

13.2.4 AUDITIONING A MIX

In order to be heard other than when SOLO/PFL-ed, channels MUST be routed to the MAIN MIX bus. This can be either from the Channel routing matrix directly, via one of the subgroups, or from the MIX-B bus (S48 DOWN).

Channels going to tape are usually monitored via the TAPE RETURN channel, whether A or B. In this case the MAIN MIX button of the channel GOING to tape should be UP, while that coming back should be DOWN (if on an A Channel). Tape tracks returning on B-Channels will always be heard so long as the MIX-B / MAIN MIX switch (S48) is depressed.

Some to-tape channels will have NO routing other than a DIRECT OUT patch into the recorder, others may be routed through GROUPS. When a GROUP is sending to tape, the GROUP L R assignment buttons (S37, 38) should be UP. I.e. that group should NOT feed into the MAIN MIX bus directly.

13.2.5 MIXER MAPPING

When laying out channels for recording or mixing, try to be sensible. Keep tom-toms together, etc.. Work out a scheme that suits you & stick to it, e.g. I always started with kick drum, snare, hi-hat, tom-toms (as the audience sees them), cymbals (ditto), bass, guitars, keyboards, other instruments, vocals. From session to session I knew what was where without hardly ever having to look at a tracksheet.

14. 8-TRACK MIDI SUITE / DANCE PRODUCTION STUDIO

8-TRACK MIDI STUDIO WITH SAMPLER, 8-TRACK RECORDING SYSTEM, ONE VOCAL MIC AND AN ARRAY OF SYNTHESISERS AND FX. SEQUENCER DRIVEN. A TYPICAL DANCE PRODUCTION SUITE.

14.1 SENDS

In a dance production, effects are often of paramount importance in creating interesting / evolving sounds, and aux sends are usually all dedicated to this purpose. Also it is not unusual for a vocalist to be drafted in to add some colour to the mix at a late stage. The following set-ups avoid using aux sends for headphone monitoring, while enabling both the vocalist and engineer to get a comfortable headphone mix.

14.2 AUXLESS HEADPHONES MIX

The vocalist will probably want to hear her-/himseif above the normal mix level. With the following Headphone sources you can do this WITHOUT tying up ANY Aux sends or MIX-B. Passing the DIRECT OUT through any 1 in 2 out delay / reverb device enables the vocalist to choose an effect she/he is comfortable singing along with.

Here the MIC CHANNEL FADER controls the AMOUNT of extra voice blended into the MAIN MIX. Adjust the FADER level FIRST until the vocalist is HAPPY, then set the level to tape with the GROUP FADER(S).

ALTERNATIVELY:

Split the mic onto 2 Channels, and use one to feed the routing matrix while the other drives the EXTERNAL input via its direct out. Now you have TOTALLY indepedent monitoring and tape send levels for the vocal signal.

ALTERNATIVELY:

If you need to hear the harmonies, but they're putting the vocalist off key, you'll want to be able to delete channels from the headphone mix. If you still don't want to assign a couple of aux buses to headphones monitoring because this would disturb the MAIN MIX, the following suggestion might prove useful. Set up a separate channel assignment on a spare Group pair. Feed the output into Aux return 3. Route Aux Return 3 to Headphones 1 (S 55) and de-assign the MAIN MIX (S77). Now you can delete distracting channels from the vocalist's backing track mix (see fig. 14.4).

This configuration does not allow for anything coming in on B-Channels to be sent to headphones. If you need to do this, S76 must also be depressed.

In all cases the WET/DRY balance of the EXTRA vocal signal takes place WITHIN the FX processor.

The above example refers to Aux 3 return > Headphones 1. An analogous situation is possible for Aux 3 or 4 Return > Headphones 1 or 2.

14.3 RETURNS

Channels 1-8 : When you've only got 8 tape returns, you can afford to bring them back on main channels to enable e.g. chorus vocal comps. or recorded real-time mixing effects such as frequency sweeping to be quickly bounced or sampled off via GROUPS.

Channels 9-23 : The MOST IMPORTANT SYNTH/SAMPLER outputs. Those most likely to need full EQ or to be recorded to tape. You might have one minimoog, but half-a-dozen uses for it. Put it on an A-channel. You'll want to record and/or sample it in action.

Channel 24 is of course the MIC input. A compressor might be patched into the channel 24 insert. Keep this channel free until the mix absolutely demands its services, just in case you want to add in any last minute singing, or any last minute anything!

The B-channel line inputs (tape returns) can accomodate even more MIDI expanders and synths etc.

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14.4 LINING UP RECORD / SAMPLE INPUTS

Set the relevant TAPE OUTPUT and INPUT switches (located at the rear of the console) to match the operating level of your 8-track (consult manual, 'phone manufacturer, or simply 'suck & see' to find which setting works best). The sampler's variable input gain range should be more than wide enough to accept either -10 or +4 dB. There is no OSCILLATOR in the EURODESK, but you can use a simple

unmodulated sustained tone from a keyboard. Choose one around 1 kHz (B above middle C is 997 or 1002 Hz depending on whether you are using the tempered scale or 'just' tuning: either way it's close enough for jazz). Set the channel EQ to OFF, and line up the channel according to the SETTING UP PROCEDURE (Section 13,1). Route this signal to all groups and adjust the GROUP OUTPUT FADERS until the bargraph meters read 0 dB. Now put the recorder into INPUT mode on all channels, and the sampler into SAMPLE mode. If the tape operating level switches are correctly set, then 0 dB on the group output meters should also show OdB on the tape recorder's input meters. A discrepancy of +/-14 dB indicates a wrong operating level selection. Small discrepancies MAY be taken up by the GROUP FADERs, though a better solution would be to get the MULTITRACK, properly aligned. (Refer to multi-track manual and/or qualified personnel.) Adjust the sampler's input level until it also reads OdB.

Beware of inaccurate/uncalibrated sampler input meters. Work out how hard you can ' safely drive the sampler's input, reference this to OdB on a EURODESK group meter, then take note of the sampler's input gain pot setting. (Or use soft adhesive tape etc. to hold it in one position.) (For more info on digital metering and associated problems see Section 13.2.3.)

14.5 MIXDOWN

The situation here is no different from record, really, save that the GROUPS may now be routed directly to the MAIN MIX (L/R) bus (S35-38) for easier mixing. Remember, you started off with the tape returns coming up on A- Channels 1-8, therefore there is no need to 'flip' them. You will probably (definitely) be running lots of sequencer tracks live. Take care not to encourage MIDI delays (see Section 22: SEQUENCING 'LIVE').

15.16-TRACK RECORDING WITH 2 SAMPLERS

15.1 RECORDING

Tape outputs 1-8 (1st row) and 1-8 (2nd row) should be wired to the multitrack record inputs 1-16. Sampler inputs should be connected to group outs 5, 6, 7 and 8 on the third row. Lining up is as per the previous example (see Section 14.3). When choosing which outputs to assign where, you have to consider that you have got maybe 16 tape and 16 or 20 sampler outputs to accommodate onto 24 A channels (and 24 B-channels)! Which are you most likely to want to bounce FROM? I'd suggest all audio tape tracks be returned on A-Channels, while at least one stereo output from a sampler is also brought back on a pair of A-Channels for 'flying in'(a sampler can pick up e.g. chorus vocal and drop it into all choruses, or sample a particularly nifty bit of flanging on a drum loop etc.; then lay the effected loop back to tape, without re-patching). Most other sampler outputs and MIDI keyboards which need to be heard but not recorded can be assigned to B-Channels. The remaining 6 or so A Channels may then be used for overdubs.

15.2 HEADPHONES

While auxless headphone monitoring (see Section 14.2) is still an option (and a bloody good one), a small general purpose studio might require a more straightforward way of working using one or two discrete headphone mixes. Here it would be best to keep Aux sends 1 and 2 free for monitoring purposes until mixdown time.

Aux returns 3 and 4 can be routed directly to headphones 1 and/or 2. A good idea would be to drive HP1 from a combination of AUX RETURN 3 (355) and MIX-B (S76), while HP2 picked up it's signal from AUX RETURN 4 (S69) and MIX-B (S88). Channel Auxes 3 and 4 would be routed to A Channels. In this configuration a reasonable degree of balancing between the MIX-B and Aux 3/4 level into the cans is possible by adjusting P55/68 (-infinity to +15 dB).

With the headphone configuration shown below, there is no easy way to get FX returns into the cans. Bring these back on A or B channels instead, until mixdown time.

15.3 MIXDOWN

With 24 A-channels and up to 36 significant tape and sampler TRACKS to accommodate, some thought will need to be given to mixdown assignments. Tracks which need little EQ and no access to the main track reverbs/echoes on auxes 1 and 2 may be parked on B-Channels. Lead tracks and prominent rhythm/melodic voices should be placed onto A-Channels. Remember, if Auxes 3/4/5/6 are routed to a B-Channel, they are denied to that strip's A-Channel, and vice versa.

16. **PROFESSIONAL 24-TRACK STUDIO**

16.1 RECORDING

Chances are you'll occasionally want to record more than eight tracks at once, e.g. it you're recording a band playing together live. The following example covers a rock band with drums, bass, 2 guitars, percussionist, brass section, lead and backing vocals. In the real world, you'll hardly ever be taking all these artists at once, but if you are:

Channels ***Route*** ***Destination***

1 Kick——————————Direct Out—-—-—Track 2

2 Snare————————Direct Out————Track 3

3 Hi Hat—————————Direct Out————Track 4

4 Tom 1—————————Group 5 + 6-———Tracks 5+6

5 Tom *2*————- ———Group 5 + 6-———Tracks 5 + 6

6 Tom 3—————-———Group 5 + 6-———Tracks 5+6

7 Tom 4————————~-Group 5 + 6-———Tracks 5 + 6

8 Cymbals (Overheads) L——~-DirectOut————Track 7

9 Cymbals (Overheads) R——~-DirectOut————Track8

10 Bass mic———————~-Group 3————Track 1

11 Bass Dl————--————Group 3————-Track 1

12 Escaping prisoners————--Group 4————-Who Knows?

13 Trumpet———————~-Group 1 + 2———Tracks9 + 10

14 Trombone——————~-Group 1 + 2———Tracks9 + 10

15 Sax————————Group 1 + 2———Tracks 9+10

16 BVs 1—-——————Group 7 + 8-———Tracks 11 + 12

17 BVs2—~——————Group7 + 8-———Tracks 11 + 12

18 BVs 3——————————Group 7 + 8———Tracks 11 + 12

19 Conga L————————Direct Out————Track 13

20 Conga R~-—————Direct Out————Track 14

21 Guitar 1 mic——————— —Direct Out— ——Track 15

22 Guitar 2 mic-—————Direct Out— ——Track 16

23 Lead Vocal——————Direct Out————Track 17

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Table 16.1: Channel assignments

Tape Monitoring will be via MIX-B in TAPE return mode. Once tracks are layed, they will be FLIPPED onto the A CHANNELS (1-17), and overdubbing can commence via tracks 18-24. Also 'bouncing', reducing several tracks onto one or a stereo pair, requires access to the full routing matrix. This is available to the A Channels, but not the B-Channels.

16.2 VERY TRICKY HEADPHONES

In a multi-musician scenario you'll probably want as many different headphones mixes as you can muster. With the current channel/group/track assignments it is possible to set up 4 independent, or semi- independent, headphone feeds while still keeping back auxes 3/4/5/6 for 'wet' monitoring (see 16.3).

Group 7—- Aux return 3 —>- HP1 Group 8 —- Aux return 4 -ᵗ HP2 MAIN MIX and/or MIX-B — HP1 MAIN MIX and/or MIX-B -»• *HP2 ACTIVE SWITCHES: S55/S75/S76 ACTIVE SWITCHES: S69/S87/S88*

(See also fig. 14.4 'Group driven auxless headphones mix' & associated text.)

In the above configuration HP1 & HP2 comprise a blend of the MAIN MIX, MIX-B and an additional feed from a group. The group feed can be used to select which channel(s) should be BOOSTED in the cans relative to the main mixes.

Choose to send to Groups 7 & 8 from channels which are routed to tape from their DIRECT OUT. Otherwise CHANNEL PAN, which will already have been set for recording via another group, is unlikely to be pointing to where you want the cue signal to go. Aux sends 1 and 2 are available as two separate MONO headphone feeds, or as a single stereo headphone feed. You'll need an extra stereo amplifier to amplify the Aux 1 & 2 outputs to drive headphones properly.

OVERDUBBING:

It's all change. Flip the recorded music onto A-Cnannels, in order to feed from tape into all the headphones buses.

ALTERNATIVELY

(and probably much more sensibly): use the headphones routine outlined in Section 15.2 (Fig 15.1).

16.3 WET MONITORING

It is customary with live recording to lay tracks DRY. (Not so with MIDI set-ups: often a tape track is used to RECORD a complex EFFECT. In a MIDI studio a TAKE is generally sequencer driven, and hence reproducible should the recorded effected track eventually prove to be unsuitable, In live recording, a great TAKE is irreplaceable! Hence the extra caution when laying live tracks.) With dry recording you will probably want to audition tape tracks with some reverb and/or echo, to get a better idea of how the final MIX might sound. By pressing the AUX 3/4/5/6 SOURCE switch (23), aux buses 3 and 4 are available to the B, i.e. Tape monitoring, Channels. (You COULD send to reverb from INPUT channels, but the FX would disappear on tape playback.) Bring the FX back on Aux Returns 1, 2, 5 or 6. (Remember 3 & 4 have been used for Headphones patching.)

16.4 MIXDOWN

All aux sends and groups are now available for mixing, as are A-Channels 23 + 24.

TIP: If you have two different instruments recorded onto one track, the MIXDOWN settings for each might be totally different. Set up two A-Channels. one for each instrument, and SWITCH between them.

The B-Channels may be used e.g. as FX returns in place of the normal Aux Returns (the advantage being that these channels have PAN and EQ), OR as an extra stereo AUX send.

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8. WARRANTY

§ 1 WARRANTY CARD

To be protected by this warranty, the buyer must complete and return the enclosed warranty card (signed/stamped by retail dealer) within 14 days of the date of purchase to BEHRINGER INTERNATIONAL (address see § 3). Failure to return the card in due time (date as per postmark) will void any extended warranty claims.

§ 2 WARRANTY

1. BEHRINGER INTERNATIONAL warrants the mechanical and electronic components of this product to be free of defects in material and workmanship for a period of one (1) year from the original date of purchase, in accordance with the warranty regulations described below. If any defects occur within the specified warranty period that are not caused by normal wear or inappropriate use, BEHRINGER INTERNATIONAL shall, at its sole discretion, either repair or replace the product.

2. If the warranty claim proves to be justified, the product will be returned freight prepaid by BEHRINGER INTERNATIONAL within Germany. Outside of Germany, the product will be returned at the buyer's expense.

3. Warranty claims other than those indicated above are expressly excluded.

§ 3 RETURN AUTHORIZATION NUMBER

1. To obtain warranty service, the buyer must call BEHRINGER INTERNATIONAL during normal business hours BEFORE returning the product (Tel.: +49 (0) 21 54 / 92 06 66). All inquiries must be accompanied by a description of the problem. BEHRINGER INTERNATIONAL will then issue a return authorization number.

2. The product must be returned in its original shipping carton, together with the return authorization number, to the following address:

BEHRINGER INTERNATIONAL GmbH
Service Department

Hanns-Martin-Schleyer-Str. 36-38

D - 47877 Willich-Münchheide

3. Shipments without freight prepaid will not be accepted.

§ 4 WARRANTY REGULATIONS

1. Warranty services will be furnished only if the product is accompanied by an original retail dealer's invoice. Any product deemed eligible for repair or replacement by BEHRINGER INTERNATIONAL under the terms of this warranty will be repaired or replaced within 30 days of receipt of the product at BEHRINGER INTERNATIONAL.

2. If the product needs to be modified or adapted in order to comply with applicable technical or safety standards on a national or local level, in any country which is not the country for which the product was originally developed and manufactured, this modification/adaptation shall not be considered a defect in materials or workmanship. The warranty does not cover any such modification/adaptation, irrespective of whether it was carried out properly or not. Under the terms of this warranty, BEHRINGER INTERNATIONAL shall not be held responsible for any cost resulting from such a modification/adaptation.

3. Free inspections, maintenance/repair work and replacement of parts are expressly excluded from this warranty, in particular if caused by inappropriate use. Likewise, the warranty does not cover defects of expendable parts caused by normal wear of the product. Expendable parts are typically faders, potentiometers, switches and similar components.

4. Damages/defects caused by the following conditions are not covered by this warranty:

▲ misuse, neglect or failure to operate the unit in compliance with the instructions given in the user or service manuals.

▲ connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used.

▲ damages/defects that are caused by force majeure or by any other condition beyond the control of BEHRINGER INTERNATIONAL.

5. Any repair carried out by unauthorized personnel will void the warranty.

6. Products which do not meet the terms of this warranty will be repaired exclusively at the buyer's expense. BEHRINGER INTERNATIONAL will inform the buyer of any such circumstance. If the buyer fails to submit a written repair order within 4 weeks after notification, BEHRINGER INTERNATIONAL will return the unit C.O.D. with a separate invoice for freight and packing. Such cost will also be invoiced separately when the buyer has sent in a written repair order.

§ 5 WARRANTY TRANSFERABILITY

This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BEHRINGER INTERNATIONAL.

§ 6 CLAIM FOR DAMAGES

Failure of BEHRINGER INTERNATIONAL to provide proper warranty service shall not entitle the buyer to claim (consequential) damages. In no event shall the liability of BEHRINGER INTERNATIONAL exceed the invoiced value of the product.

§ 7 OTHER WARRANTY RIGHTS

This warranty does not exclude or limit the buyer's statutory rights provided by national law, in particular, any such rights against the seller that arise from a legally effective purchase contract.

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