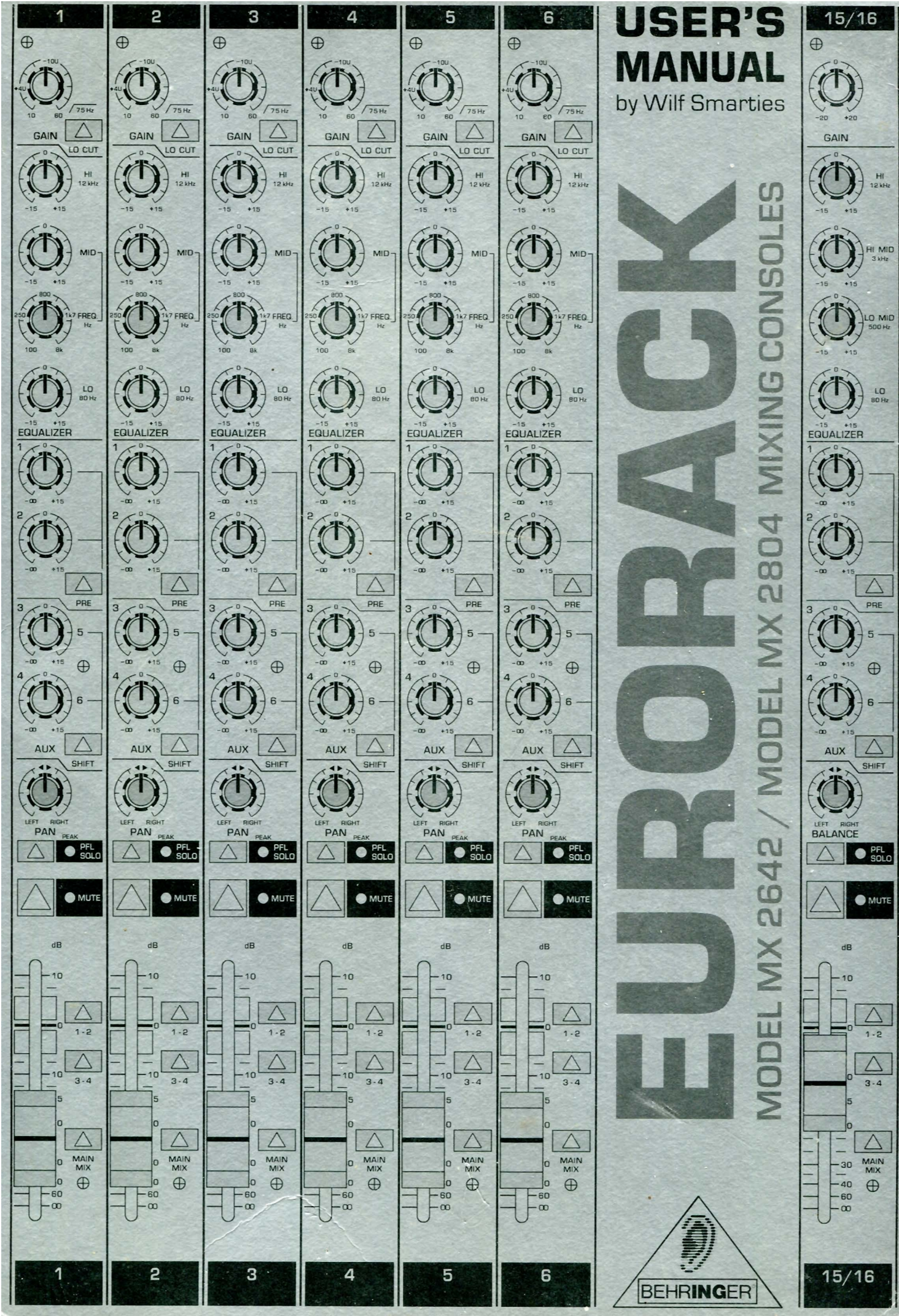
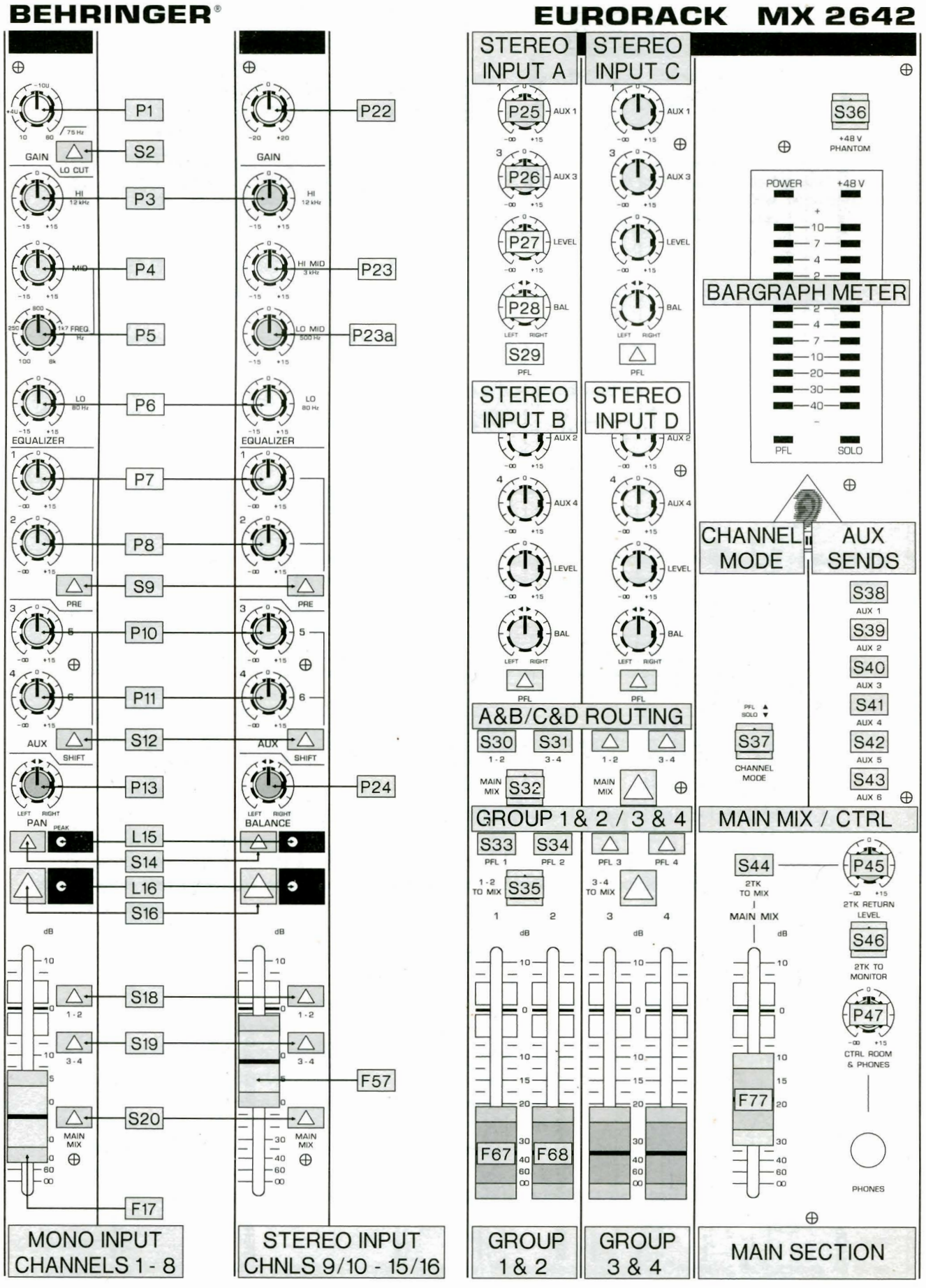
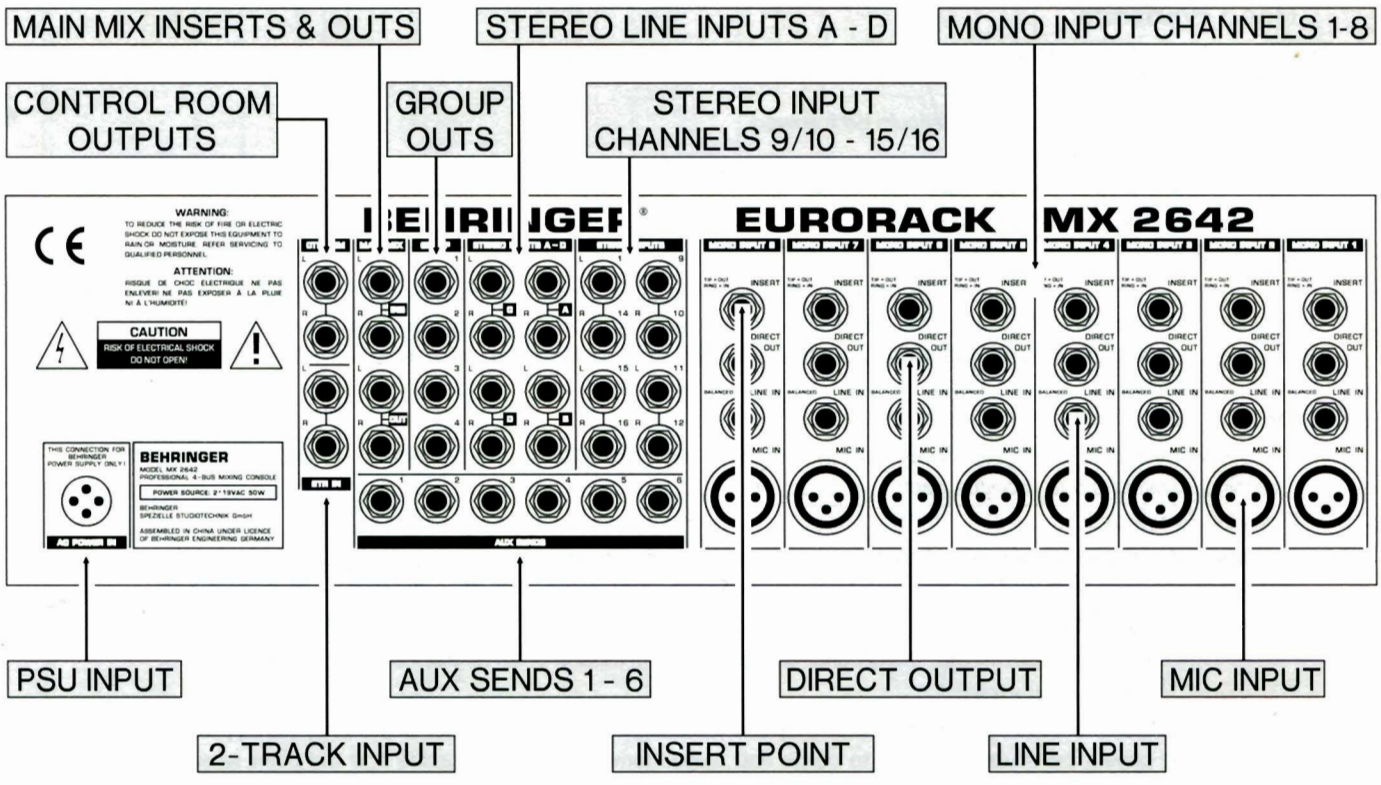

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2
3
4
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15/16
USER'S MANUAL
by Wilf Smarties
EURORACK
MODEL MX 2642 / MODEL MX 2804 MIXING CONSOLES
GAIN
LO CUT
HI
12 kHz
MID
1k7 FREQ
LO
80 Hz
EQUALIZER
PRE
AUX
SHIFT
PAN
PEAK
PFL SOLO
MUTE
dB
1-2
3-4
MAIN MIX
HI MID
3 kHz
LO MID
500 Hz
BALANCE
BEHRINGER

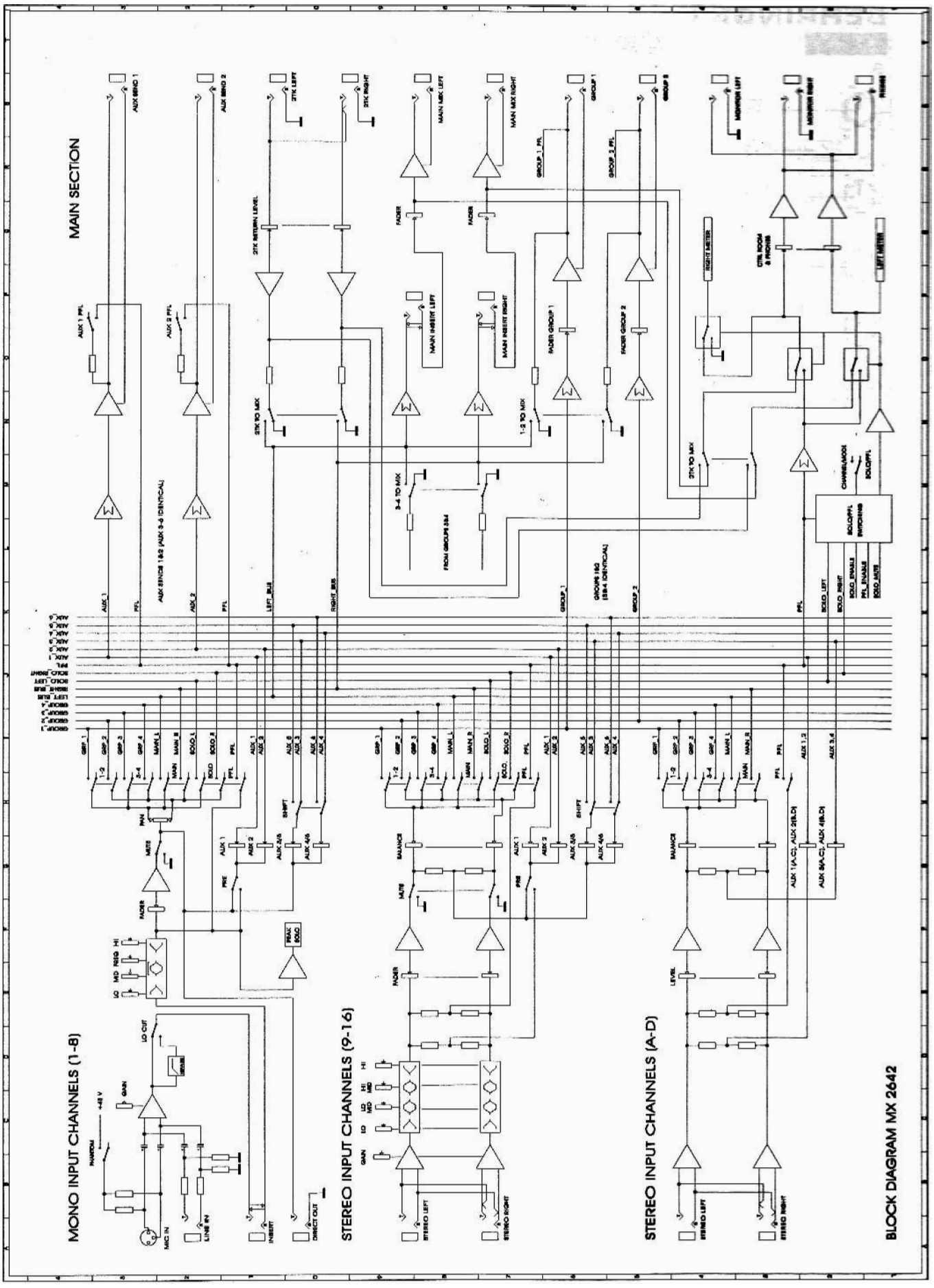
MX 2642 FRONT PANEL



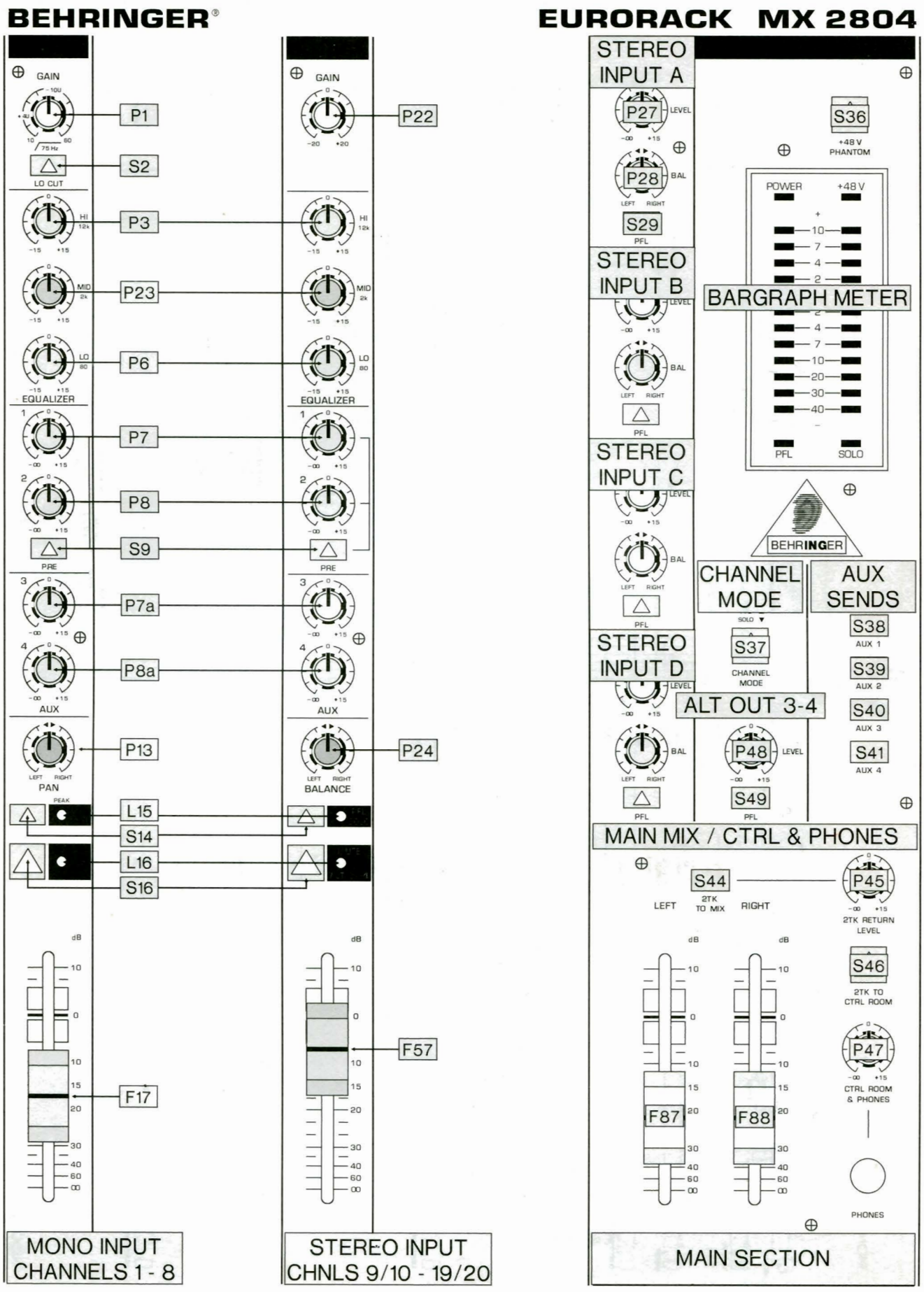
MX 2642 CONNECTOR PANEL



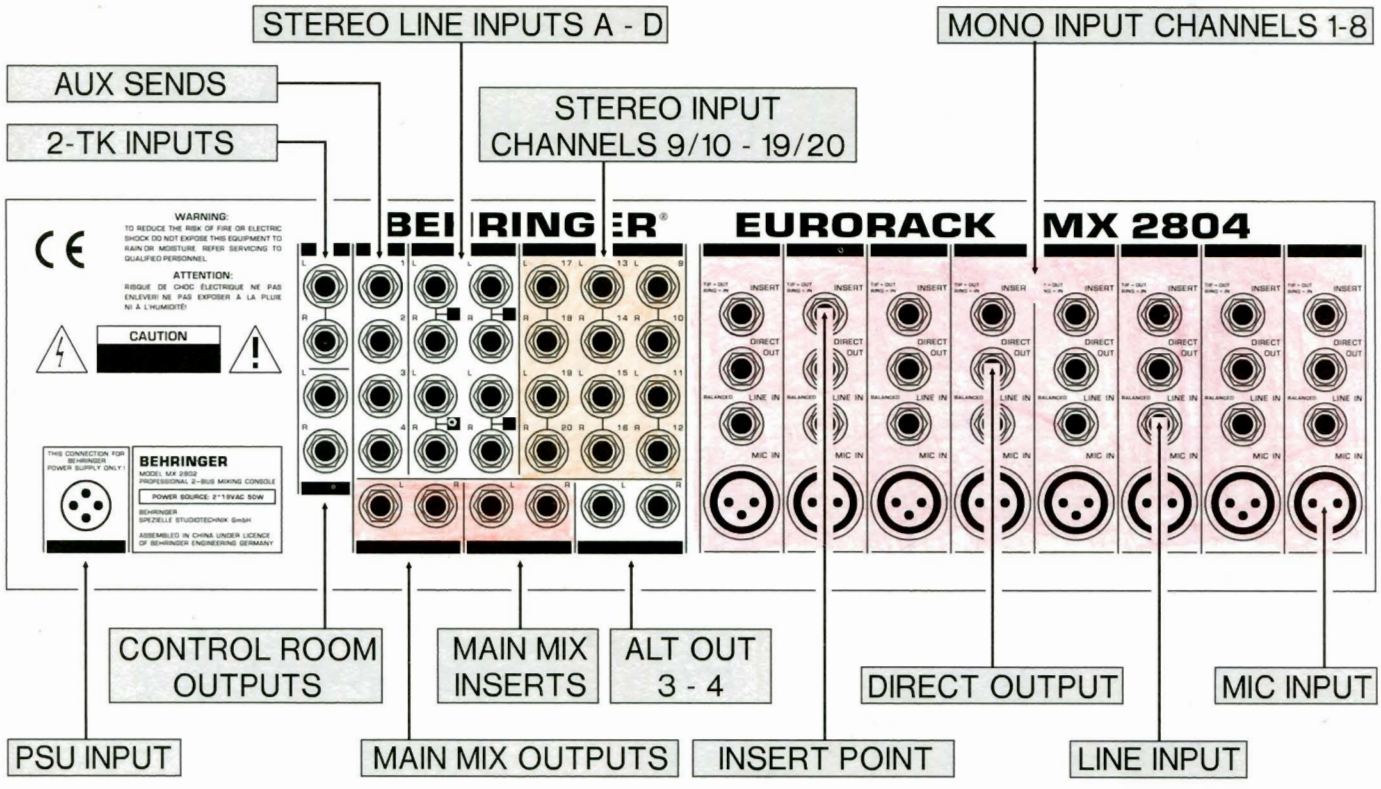
MX 2642 BLOCK DIAGRAM



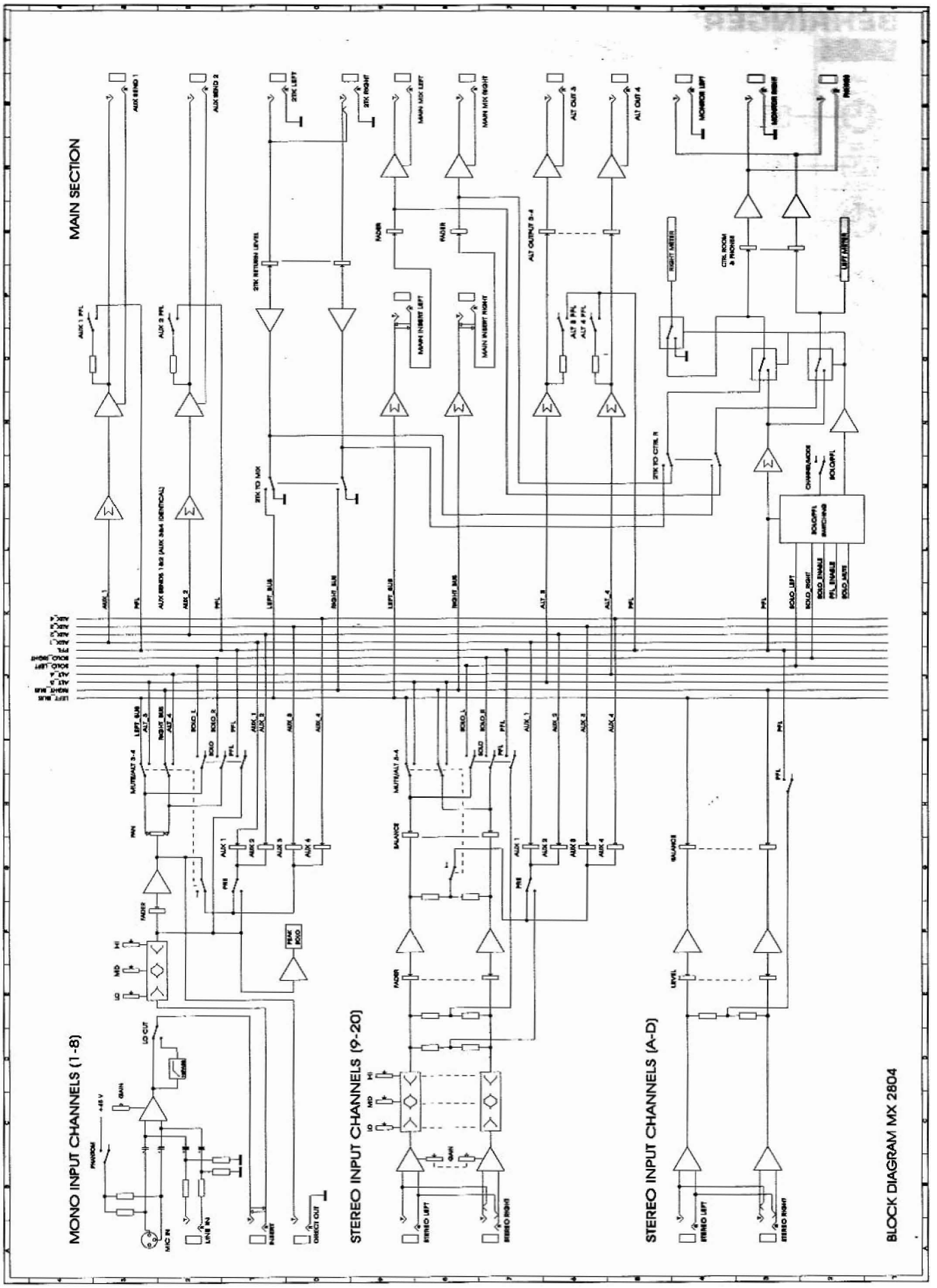
MX 2804 FRONT PANEL



MX 2804 CONNECTOR PANEL



MX 2804 BLOCK DIAGRAM



WELCOME

Dear Sound Engineer,

Welcome to the Behringer EURORACK and thankyou for placing your trust in Behringer products. My most pleasant task is to write this letter to you, because it is the culmination of many months of hard work for our engineering team. Their daily objective is to be focused on you, the Sound Engineer, and with that focus in mind, it drives us to reach a goal which is unique, and is the backbone of the Behringer philosophy.

That philosophy is very simple, that you the Sound engineer, are the most important part of the Behringer family. We will not accept anything less for you than the highest quality, best specified product, at the lowest price. This enables you to express your full creativity without price being a concern.

Due to the thousands of satisfied Behringer users around the world, we are able to demand the highest quality components industry can offer, at extremely low prices, making the final cost of our products astonishingly low. Why then, shouldn't we pass the benefit of our good fortune back to you, the Sound Engineer? This enables you to own the type of quality equipment that previously would have cost very much more. It also allows you the confidence that your new equipment uses technology which is leading the way in the field of Audio design.

I would like to thank the following people, whose help on 'Project EURORACK' has made it all possible:

* The existing users of Behringer equipment (whose comments and suggestions have made them the most important members of the Behringer design team),
* Bettina Liesen (for this marvellous manual layout),
* Bernhard (Rammi) Ramroth (whose technical ingenuity is unique),
* Frank for the fine mechanics (key-word 'Tooling modification') and
* Wilf Smarties (for his immeasurably helpful work on this manual).

Can you imagine the feeling of mixing a gold record? We believe a creative composer lies in you and we want to share in your success. Our products are built for people like you -people with a fine ear- and as you have trusted us by buying the Behringer EURORACK, we return your trust by welcoming you into the Behringer family.

Thankyou and sincerely yours,



BEHRINGER Spezielle Studiotechnik GmbH

Uli Behringer

President

EURORACK

26 INPUT 4-BUS MIXING CONSOLE
MODEL MX 2642
28 INPUT 4-BUS MIXING CONSOLE
MODEL MX 2804

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1.0 THE MANUAL

1

1.1 INTRODUCING THE EURORACK

Congratulations. In purchasing our EURORACK you have acquired a mixer whose small size belies its incredible versatility and superlative audio performance. Your EURORACK is built to the same outstanding quality as our top-of-the range console, the Behringer EURODESK MX 8000.

MX 2642

offers you a choice of Mono and Stereo Input Channels, four assignable Subgroups, and four Stereo Line Inputs.

MX 2804

offers you a choice of Mono and Stereo Input channels, plus four Stereo Line Inputs.

Later we will see how adding the Behringer ULTRALINK to your system can inexpensively expand the capacity and flexibility of you mixing power even further. First of all, however, we want to take you on a guided tour of your new rack-mounted console - the Behringer EURORACK.



We recommend that you experiment with your EURORACK away from the pressures of a recording session or live concert, in order to get a feel for it. It is a musical instrument. Learn to play it well.

1.2 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. 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, try to look it up in the Glossary before you write a letter of complaint!

1.3 KEYS

Next to the front cover you will find some pages with drawings showing the front and rear panel of your EURORACK respectively. Keep them turned over, lying to the left of the text pages, while studying the manual.

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

S for Switch, L for LED, P for Potentiometer and F for Fader.

After every prefix, you will find the function number.



Passages of the text which refer to the MX 2642 or the MX 2804 only will be marked in the way you can see on top of this page.

2.0 EURORACK OVERVIEW

2

MX 2642

The configuration is 26 into 4 into 2. This means that there are 26 inputs in total (8 Mono and 4 Stereo Channels, 4 Stereo Line Inputs, and an extra 2-track Stereo Input which can also be routed to the Main Mix. 2 stereo (or 4 mono) Subgroups are provided for connecting to multitrack tape recorders, or for use as an aid during mixdown or a live concert mix.

MX 2804

The configuration is 28 into 4 (2+2). This means that there are 28 inputs in total (8 Mono and 6 Stereo Channels, plus 4 Stereo Line Inputs).

2.1 ARCHITECTURE

Mono Input Channels

Channels 1 - 8 are Mono, with a choice of balanced Mic or Line Inputs. The vintage-style high-current discrete mic amps are of the same incredible quality as those found on the acclaimed Behringer EURODESK MX 8000, while a large external power supply ensures low noise and superior transient response at all times. Insert Points and Direct Outputs are also provided, giving 'big console' functionality to the Mono Input Channels.

Stereo Input Channels

A further 8 Line Inputs on the MX 2642 (12 on the MX 2804) are configured as 4 Stereo Input Channels (6 on the MX 2804), accepting line level signals at -10 dBV (semi-pro) as well as at +4 dBu (professional) operating levels. These are ideal for multitrack tape returns, or for accepting outputs from MIDI and other electronic instruments.

Channel Outputs

A high-quality true-logarithmic 60 mm fader feeds the Main via a constant-power Channel-Pan.

On the MX 2642, Subgroup buses are fed via Subgroup and Mix Assign switches, too. Here the Channel Pan also selects between odd and even-numbered Subgroup buses.

Subgroups (MX 2642 only)

For ease and flexibility of mixing, four mono (two stereo) assignable Subgroups are provided on the MX 2642. This is the first time you have ever been offered this professional feature in a rack-mounted console.

Aux Sends

There are six Aux Send buses on the MX 2642, four on the MX 2804, each with PFL.

Stereo Line Inputs

MX 2642

Four Stereo Line Inputs, each with Solo/PFL, lie above the Subgroup faders. These inputs are each assigned to two Aux Send buses. When these are used for returning tape tracks these Aux Sends could be used as cue feeds to the performers' headphones, or as Effects Sends for 'wet' monitoring. Alternatively, the Stereo Line Inputs could be used for MIDI instruments etc.

MX 2804

Four Stereo Line Inputs, each with Solo/PFL, are sited immediately to the right of the input channels. These can be used to return stereo effects or be used as Tape Monitor Returns. Alternatively, the Stereo Line Inputs could be used for MIDI instruments etc.

Main Mix Output

MX 2642

Main Mix output level is via a high-quality true-log 60 mm stereo fader, and is monitored by a pair of accurate 12 segment bargraph peak meters, surrounded by four status LEDs. Other functions of the Main section include headphones output with gain, and a 2-Track Tape Return, which doubles as an extra stereo line input available to the mix with its own level control, completing the 26 input channels of the innovative MX 2642.

MX 2804

Main Mix output level is via a pair of high-quality true-log 60 mm stereo faders and is monitored by a pair of accurate 12 segment bargraph peak meters, surrounded by 4 status LEDs.

2.2 METERING

Channels 1 - 8 on both the MX 2642 and the MX 2804 have overload LEDs, while the L and R outputs have 12 segment bargraph meters. The L/R meters double up as mono PFL or stereo Solo meters.

The master bargraph meters should average around 0 dB during loud passages. If they read persistently higher, or are peaking above +10 dB (top segment of the main L/R LED display), reduce either the main L/R fader, the Subgroup faders (on the MX 2642 only) and/or Channel faders, or (as a last resort) channel input gain. Maybe it's time to do a round of PFL metering.

2.3 PSU (Power Supply Unit)

Any amplifier circuit is limited in its transient response by the available current. Every mixer has numerous 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 EURORACK. The sound will always stay clean and crisp right up to the operating limits of the op-amps themselves, thanks to our generous 40 W external Power Supply unit.

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

3.0 MONO INPUT CHANNEL

3

Each channel comes with a balanced Line Input on 1/4" jack, and an XLR Mic Input. Phantom powering is switchable from the Master Panel (S36). The Gain circuit has an unusually wide range from -50 dB to +10 dB, obviating the need for mic/line switching. The crucial operating levels +4 dBu and -10 dBV are clearly and accurately legended (P1).

3.1 INPUT LEVEL SETTING

Channel input level is determined by the Gain control (P1). Use Solo/PFL (S14) to bring the Channel Input onto the Left and Right bargraph meters respectively. This also sends the Solo/PFL-ed signal to the Left and Right speakers.

For level-setting (as opposed to localized listening) choose to use the mono PFL bus rather than the Post-fader (Post-channel pan) stereo Solo bus (Channel Mode Global Switch S37 UP).

Solo/PFL never interrupts the mix at the main recording outputs. It follows that Aux Sends (and Subgroups on the MX 2642) must also be unaffected, since they contribute directly to the Main Mix.

In addition to switchable Solo/PFL (S14) metering, a channel LED (L15) illuminates when a channel is going into overload. This takes its cue from three test points: Input, Post-EQ and Post-fader. This is very important, especially when using extreme EQ settings or using a dynamics processor in an Insert. In all cases the higher level wins. You do not want the overload light to come on except very intermittently during a take or a mix. If it does light persistently, reduce input gain (see also section 11: Setting Up).

There is a steep Lo Cut (High Pass) filter (S2), slope @ 18 dB/Oct, -3 dB @ 75 Hz, for reducing floor rumble, explosives, woolly bottom end etc.

3.2 EQUALIZER

All Mono Input Channels are fitted with three-band EQ and a switchable Lo Cut filter for eliminating unwanted subsonics. All three bands have up to 15 dB of cut and boost, with a centre detent for 'off'.

The upper (P3) and lower (P6) shelving controls have their frequencies fixed at 12 kHz and 80 Hz respectively. On the MX 2804, the midrange (P23) control has its frequency fixed at 2 kHz.

MX 2642 only

Here the midrange (P4) control is semi-parametric with a peaking response, Q fixed at 1 octave, sweepable from 100 Hz - 8 kHz (P5).

3.3 AUX SENDS

All Aux Sends are mono and post-EQ.

For Aux Sends 1 & 2, two dedicated pots (P7, P8) are used. These can be taken from a point before or after the channel fader, i.e. Pre or Post (S9).

MX 2642

On the MX 2642, Aux Sends 3 & 5, and 4 & 6 are serviced by two potentiometers (P10, P11). The Shift button (S12) determines whether buses 3 and 4 or 5 and 6 are addressed. Aux Sends 3/4/5/6 are always wired Post-fader.

MX 2804

On the MX 2804 Aux Sends 3 & 4 (P7a, P8a) are always wired Post-fader.

For almost all FX Send purposes, you will want Aux Sends to be Post-fader, 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-Fader, i.e. independent of the Channel Fader and Mute.

Most reverbs etc. sum internally the left and right inputs. The very few that don't may be driven in true stereo by using 2 Aux Sends.

There is +15 dB of gain on every Aux Send. Such a high boost is usually only appropriate where the channel fader is set around -15 dB or lower. Here, an almost exclusively 'wet' signal will be heard. In most consoles, such a wet mix required the use of a Pre-setting for the channel Aux Send, losing Fader control. With the EURORACK you can have a virtually wet mix with Fader control.

3.4 ROUTING, FADING AND MUTING

MX 2642 only

Routing means selecting which Bus you want a channel to address. There are 3 stereo buses in the EURORACK MX 2642 (plus a stereo Solo bus). The main L R bus is selected by (S20), while the Subgroups are selected by 1-2 and 3-4 Assign switches (S18, S19). All 4 stereo buses follow Channel Pan (P13). Usually, only one of L-R, 1-2 or 3-4 will be selected for a particular channel.

An exception to this rule is when laying down voice takes. It is often convenient to have the mic channel(s) routed to all 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 Main L R bus (and to the Group buses, if MX 2642) is ultimately determined by the Channel Faders (F17). These are made by Panasonic, and are designed to give a smooth logarithmic taper of a type more usually associated with megabuck consoles. The low level performance particularly is far smoother than that of a normal 'budget' fader.

Channel Pan (P13) positions the output of the Channel in the stereo field. Its constant-power design ensures there are no level discrepancies whether a signal is hard-panned, centre-stage, or somewhere in-between. Such pin-point accuracy will be a revelation if you have been working on consoles with lower quality circuits.

Solo/PFL we encountered in section 3.1. Solo also follows Channel Pan.

The Mute button (S16) is ergonomically placed immediately above the Channel Fader. Engaging Mute, indicated by L16, is equivalent to setting a fader level of minus infinity

MX 2804 only

On the MX 2804 the Mute button serves two functions. As long as the Alt 3-4 Outputs are not in use, S16 acts like a 'normal' Mute switch, disconnecting the Channel from the Main Mix. If you want to use the Alt 3-4 Out facility, all you have to do is connect the Alt 3-4 Outputs to whatever you want.

Useful for a lot of multitrack recording as well as live sound / mixdown purposes - see 6.7 for further information and application examples.

4.0 STEREO INPUT CHANNEL 4

Each stereo channel comes with two balanced line level inputs on 1/4" TRS jacks, for left and right signals. When only the left input is connected, the channel operates in mono.

4.1 INPUT LEVEL SETTING

Channel input sensitivity is adjustable within a +/- 20 dB gain range (S22), enabling a perfect match with most common sources including multitrack tape outputs, MIDI and other electronic instruments, and effects units, all of which are normally designed to operate at -10 dBV (semi-pro) and/or +4 dBu (pro) operating levels (see also 3.1 and section 11: Setting Up).

4.2 EQUALIZER

MX 2642

The Stereo Input Channels are fitted with four-band EQ.

The upper (P3), high midrange (P23), low midrange (P23a) and lower (P6) shelving controls have their frequencies fixed at 12 kHz, 3 kHz, 500 Hz and 80 Hz respectively.

MX 2804

Here the Stereo Input Channels are fitted with a three-band EQ.

The upper (P3), midrange (P23) and lower (P6) shelving controls have their frequencies fixed at 12 kHz, 2 kHz and 80 Hz respectively.

All bands have up to 15 dB of cut and boost, with a centre detent for 'off'.

The EQ on the Stereo Channels is in principle identical to that on Mono Channels, except that the EQ is stereo, of course!

A stereo equalizer is generally preferable to using two mono equalizers when EQ-ing a stereo signal, as often discrepancies between left and right settings can occur.

4.3 AUX SENDS

These are the same as for Mono Channels (see 3,3). Note that a mono sum is taken from the stereo input.

4.4 ROUTING

The only difference here from the Mono Channel described in 3.4 is in the implementation of the Balance Control. When a channel is run in stereo, this control functions as a Balance control, determining the relative balance of the Left and Right Channel signals being sent to the Left and Right Main Mix buses (or maybe odd and even group buses in case you've got a MX 2642). For example, with the Balance control turned fully clockwise, only the right portion of the channel's stereo signal will be routed to the Main Mix (or Alt 3-4 Bus on a MX 2804 / or to any or all of buses R, 2 or 4 on a MX 2642).

5.0 SUBGROUPS, DIRECT OUTPUTS AND INSERTS

5

MX 2642 only

5.1 SUBGROUPS

The principal routes to multitrack are via the Subgroup Outputs. There are 2 stereo (or 4 mono) Subgroups. All channels can access all of them. Subgroup level is determined by the Subgroup faders (F67, F68). PFL is supported (S33, S34).

Subgroups Outputs can also be assigned to the Main Mix in pairs during mixdown (S35). If you want to run a Subgroup in mono, it should not be fed into the Main Mix by the Main Mix switch (S35). Rather it should be brought back on a Mono Channel (or via e.g. an ULTRALINK) so that it can be panned.

If you want to insert a dynamics processor between any of the four Subgroups and the Main Mix, the group output should be used as a send, return being effected through another desk input. Here, remember to de-assign the Subgroup pair from the main mix (S35).

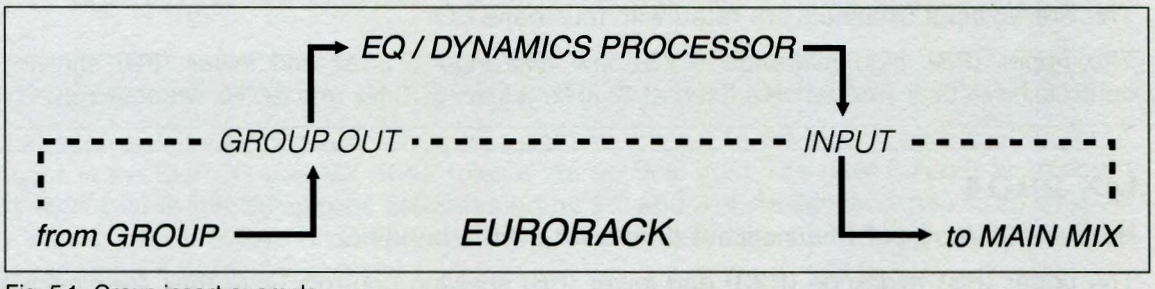


Fig. 5.1: Group insert example

The 2-Track Stereo Input is particularly suitable for returning processed Subgroup signals. Since this input cannot be routed back to any Subgroup, a feedback loop is impossible. Four Subgroup returns may easily be blended in stereo into this input simply by using the Behringer ULTRALINK MX 662 6+2 into 2 line mixer, greatly extending the functionality of your EURORACK (see section 10: Expanding).

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

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 during pauses in programme material will give the illusion that the reverb was massive all the time. (The listener will be left wondering how the singer could sound so clear in such a wet acoustic!)

5.2 MULTITRACKING

The principal routes to multitrack are via the Main Mix and Mic Channel Direct Outputs. You can also use any of the Aux Outputs to send to tape.

5.3 DIRECT OUTPUTS

Each Mono Channel on the EURORACK has its own Direct Output, which is taken from a point immediately after the fader (i.e. post EQ and after the Aux Sends). This can feed a tape track directly, enabling up to 12 tracks to be recorded simultaneously when using the Aux Sends to send to tape (up to 14 tracks in case of MX 2642 - without having to resort to using the Subgroups). Direct Outputs to tape provide the shortest possible recording signal path, and therefore highest fidelity.

5.4 INSERTS

Insert points are useful for adding dynamic processing or equalization to a channel or the mix (or group, if MX 2642). 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, fed through the dynamics processor and/or EQ, then returned to the console at the same point where it left. The insert point is normalized, i.e. the signal is only interrupted when a jack is plugged into it.

All mono input channels have got Insert Points, as has the Main Mix. Each Insert Point is accommodated on a single TRS jack socket wired tip = send, ring = return, sleeve = ground/ screen. Inserts are always Pre-fader, and also Pre-EQ/Aux sends for channels.

Insert Points may also be used as Pre-EQ direct outputs without interrupting the signal flow. You can extend the functionality of your Insert Points by wiring them onto a jackfield patchbay, where send and return can be accessed on separate sockets (see section 9: Patchfield).

Inserting into a Stereo Channel or Stereo Line Input

If you want to insert a dynamics processor etc. into any stereo input, it must be done between the source output and the EURORACK input, as these channels have no bona-fide insert point.

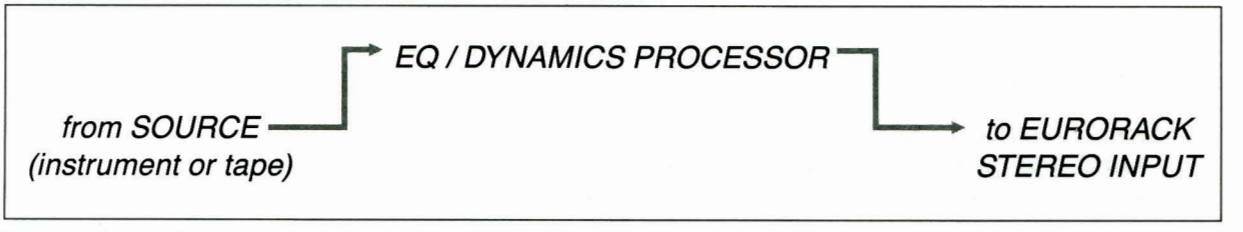


Fig. 5.2: Stereo Channel / Aux Return Insert

Post-EQ Channel Insert

If you want to insert a dynamics processor POST-EQ, the Direct Output should be used as a send, return being effected though another desk Channel, or possibly via a separate line mixer (such as the Behringer ULTRALINK), patched into e.g. the 2-Track or any other suitable stereo return channel.

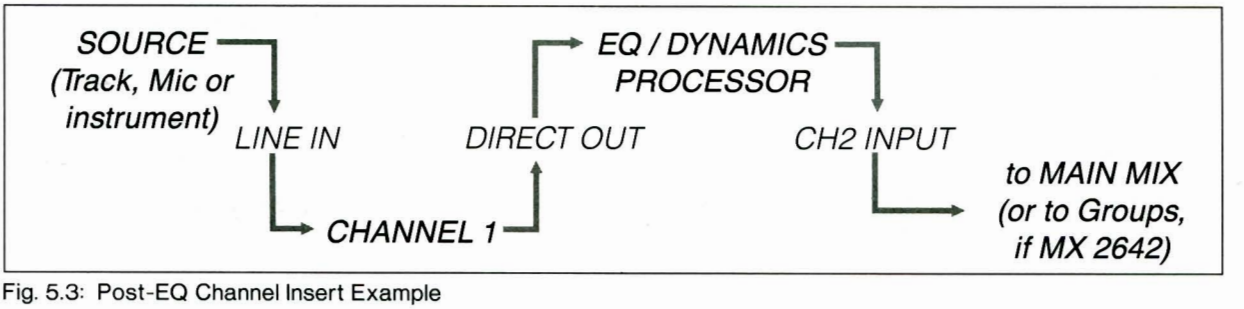


Fig. 5.3: Post-EQ Channel Insert Example

In this arrangement you might find that Compression tends to soften the perceived amount of (especially treble) EQ applied. The solution? Apply more EQ. This creates a real 'pressure' sound, great for high energy music such as dance.

6.0 MAIN SECTION

6

6.1 AUX SENDS

These are on 1/4" balanced jacks at +4 dBu. No master Aux Send level pots are provided. However, you can use the input level control of your effects unit to match the output level from the EURORACK. If your effects unit does not have an input gain, then remember that every channel and Aux Send has up to 15 dB gain, which should be more than enough to drive any effects unit. All Aux Sends (6 on the MX 2642, 4 on the MX 2804) have a PFL facility (S38-43 on the MX 2642, S38-41 on the MX 2804) to check that the Aux Send buses are being comfortably driven from the Input Channels and Subgroups.

6.2 STEREO LINE INPUTS

There are four stereo Aux Line Inputs, A, B, C and D. These operate in mono if only the Left Channel socket has a jack connected.

Each has Level (P27), Balance (P28), and PFL (S29). These Line Inputs have been cleverly designed to be multi-functional as described below.

MX 2642

Stereo Line Inputs can send to two of the Aux send buses (Auxes 1 and 3 in the cases of A and C, and Auxes 2 and 4 in the cases of B and D).

Tape Monitor Returns

You could use the Stereo Line Inputs as tape returns from a multitrack recorder.

MX 2642

You might want to feed tape tracks into the artists headphones. Using Aux sends 1 or 2 as a Cue Feed you can do this. You might want to 'wet monitor'. Using Aux Sends 3 or 4 you can add reverb or echo without tying up a main desk channel.

Instrument Inputs

The Stereo Line Inputs may be used as extra instrument inputs, especially if your MIDI keyboard or rack supplies a pre-mixed stereo signal

MX 2642

When using the Stereo Line Inputs as extra instrument inputs, you have access to the Aux buses for effects processing.

Effects (Aux) Returns

The Stereo Line Inputs may be used for returning the outputs of effects units.

MX 2642

Often it is useful to be able to send an effect into another effect. Echo sounds fatter when put through a stereo chorus or phaser. To do this use the Aux Send facility of the Stereo Line Inputs. Just be sure not to send to an effect from its own output, creating feedback.

Certain stereo effects produce a perceived imbalance between the left and right channel levels.

When applying short left and right delays, the shortest one will always seem loudest. When pitch shifting up and down in wide stereo to thicken a sound, the signal shifted upwards will seem louder than one that goes down. In both cases use the Pan (Balance) control (P28) to compensate. (An analogy comes from Greece: the columns of the Parthenon in Athens are slightly bowed so as to appear straight.)

When carrying any 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, just in case you have any significant hearing discrepancies between your ears (and if you are in the habit of listening to high sound pressure levels (SPLs), chances are you will).

Sometimes an engineer wants to narrow the stereo width of a reverb field. To do this you will have to come back on two mono channels to get independent Pan for the left and right signals.

MX 2642 only

Stereo Line Input Assignment

Each of A + B and C + D is assignable to all or any of Main Mix (S32), or Subgroup pairs 1+2 and 3+4 (S30, S31). Normally these will be assigned only to the Main Mix. However, provision is made for routing to the subgroups, in case you want to record the signal to tape, or mixdown via a Subgroup.

6.3 METERING

Main Mix/Solo/PFL level is displayed on a pair of highly-accurate 12 segment bargraph Peak meters. LEDs indicate Power On (top left), +48 V Phantom Power present (top right), and whether the mono Pre-Fade Listen bus (bottom left) or the stereo Solo bus (bottom right) is engaged.

6.4 CHANNEL MODE

The Channel Mode switch (S37) determines whether Solo-In-Place or Pre-Fader Listen is assigned to the Channel Solo buttons.

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 Sends, Stereo Line Inputs and, in case you got a MX 2642, Subgroups, too. The Solo bus is always Post-fader.

PFL

Pressing S37 once disengages the stereo Solo bus, and replaces it with a separate mono PFL (Pre-Fade Listen) bus. Now any channel which is Solo-ed, isn't. It is PFL-ed instead. PFL should always be used for gain-setting (see also the essential section 11: Setting up).

6.5 2-TRACK INPUT AND OUTPUT

Input

The 2-Track Input is on balanced 1/4" TRS jacks at +4 dBu, and is primarily for auditioning mix playback from tape. S46 routes this signal to the monitors. However, it can also be routed to the Main Mix via S44. Here S46 should be disengaged, or you will be listening to the 2-Track signal twice over! With S44 depressed you have another Stereo Line Input available to the mix, suitable for accepting the output of a second EURORACK or the Behringer ULTRALINK (see section 10: Expanding).

With the level control (P45) fully clockwise, your 2TK-Input will be matched to the semi-professional level -10 dBV. For higher output recording sources (eg. +4 dBu) turn the level of P45 down.

Output

A single pair of balanced jacks deliver the Mix Output to your 2-Track recorder (or PA system) at +4 dBu. Level to tape is ultimately determined by a precision Stereo Fader (F77). A Mix Insert Point is provided for patching a gate, compressor etc. pre-fader. This is important: connecting a compressor or gate after the 2-Track output would disrupt any attempt at a smooth fade using the output fader (F77). Although 2-Track Output is primarily designed for recording, it can also be used as a PA feed, or as a send to the input of your sampler.

6.6 MONITORING

A single volume control sends the level to the headphones and main monitors (P47).

The L/R meters follow whatever source is being auditioned (the meters won't make much sense if more than one source is selected!)

Selecting Solo/PFL (or monitoring a source other than the Main Mix on the MX 2642) 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!

Though most of you will want to audition the Main Mix most of the time there are exceptions. These include Solo/PFL of course, and also 2-Track (or other external source) playback. The 2-Track Input could be 'normalled' to a hi-fi pre-amp, allowing you to monitor extra sources such as vinyl, cassette, CD etc.

MX 2804 only

6.7 ALT 3-4 OUTPUT

Level to the Alt 3-4 Outputs is adjusted by P48. Use S49 to have the Signal PFLed.

Alt 3-4 is likely to bewilder newcomers. We want to give you two examples for the use of this feature.

When recording to multitrack, use the Alt 3-4 Outs to feed the multitrack. With most multitracks, you can multiply the Alt 3-4 Outs, using Y-leads or mults, to feed multiple tracks. Take Alt Out Left and send it to the odd tracks, and Alt Out Right and send it to the even tracks. Now tracks in Record or Input mode will hear the Alt 3-4 signals, tracks in Playback or Safe mode will ignore them. Return to your MX 2804 via an unused Stereo Line Input to make sure what's going to tape.

Sometimes you may want to control the level of several channels with one knob. Time for Subgrouping via the Alt 3-4 Outs! Assign the channels you want to the Alt 3-4 Outs, then patch back into an unused Stereo Channel.

Everytime you go back from the Alt 3-4 Outs into a Stereo Channel, do not engage the Mute/Alt 3-4 button on this Stereo Channel (this will result in a feedback, of course)!

7.0 QUESTIONS & ANSWERS 7

7.1 MULTITRACKING

MX 2642

Q?: Can I use the MX 2642 mixer with my eight-track recorder?

A: Yes. There are a number of ways to achieve this. If you want to record more than four separate tracks at once, you could use the 4 Subgroups to send to 4 record tracks, and/or any of the 8 Direct Channel Outputs to send to other tape tracks (eg for solo instruments like Bass Guitar, Kick Drum etc.). You could also use the Aux Sends as (up to) 6 mono or 3 stereo submixes to tape. In fact, you could record 18 unique tracks simultaneously! Of course, it is not anticipated that you'd want to do this from a compact console like the MX 2642!

SUBGROUPS TO TAPE

It is much more likely that, if you are laying to tape or hard disk, you will be doing so one, two or at the most 4 tracks at a time. In which case you have sufficient Subgroups to handle all the to-tape audio traffic. In order to access more than 4 tape tracks from these Subgroups, we recommend that you parallel wire tape tracks 1, 5, (9, 13 etc.) to Subgroup output 1, tracks 2, 6, (10 etc.) to Subgroup 2, and so on. This 'multiple bussing' is best done on a patchbay. (Please note, in the example given in Section 9 Patchfield, this was not done. Another configuration was chosen.) Subgroups are a very convenient route to tape, particularly when recording successive overdub 'takes'.

DIRECT TO TAPE

If you require to record more than 4 tracks at a time, or if you want to use the shortest possible signal path (for optimum quality), then you will want to use at least some of the Direct Outputs. (A mixture of Subgroup and Direct Outs is commonly used when multitracking).

MONITORING FROM TAPE

Tape Returns may be via any line input of the EURORACK. A stereo submix of, for example, drum kit or backing vocals is probably best brought back on a Stereo Channel. Find out the right Input Gain by turning P22. A mono track is best assigned to a Mono Channel. -10 dBV or +4 dBu is clearly marked on the Mono Channel's Gain pot (P1).

MX 2804

Q?: Can I use the MX 2804 mixer with my eight-track recorder?

A: Yes. There are a number of ways to achieve this. If you want to record more than four separate tracks at once, you could use the 4 Aux Sends to send to 4 record tracks, and/or any of the 8 Direct Channel Outputs to send to other tape tracks (e.g. for solo instruments like Bass Guitar, Kick Drum etc.).

AUX SENDS TO TAPE

It is likely that, if you are laying to tape or hard disk, you will be doing so one, two or at the most 4 tracks at a time. In which case you have sufficient Aux Sends to handle all the to-tape audio traffic. In order to access more than 4 tape tracks, you could parallel wire tape tracks 1, 5, (9, 13 etc.) to Aux Sends 1, tracks 2, 6, (10 etc.) to Aux Send 2, and so on. This 'multiple bussing' is best done on a patchbay (see section 9: Patchfield).

DIRECT TO TAPE

If you require to record more than 4 tracks at a time, or if you want to use the shortest possible signal path (for optimum quality), then you will want to use at least some of the Direct Outputs.

MONITORING FROM TAPE

Tape returns may be via any line input of the EURORACK. A stereo submix of, for example, drum kit or backing vocals is probably best brought back on a Stereo Channel. Find out the right Input Gain by turning P22. A mono track is best assigned to a Mono Channel. -10 dBV or +4 dBu is clearly marked on the Mono Channel's Gain pot (P1).

7.2 TALKBACK

MX 2642

Q: How do I communicate with performers in my studio?

A: As there is no talkback mic in the EURORACK, you will have to use a mic channel and a separate microphone, routed via the cue feeds to the artists headphones. These would normally be taken from Aux Sends 1 and 2 (P7, P8), set to Pre-Fader (S9). The Behringer POWERPLAY Headphone Distribution Amplifier could easily provide the basis for four stereo powered headphones feeds for your studio. Note that the Stereo Line Inputs, when used as Tape Monitor Returns, are configured assuming Auxes 1 and 2 to be the main (mono) cue feeds.

MX 2804

Q: How do I communicate with performers in my studio?

A: As there is no talkback mic in the EURORACK, you will have to use a mic channel and a separate microphone, routed via the cue feeds to the artists' headphones. These would normally be taken from Aux Sends 1 and 2 (P7, P8), set to Pre-Fader (S9). The Behringer POWERPLAY Headphone Distribution Amplifier could easily provide the basis for four stereo powered headphones feeds for your studio.

8.0 CONNECTIONS

8

8.1 RELOCATING THE CONNECTOR PANEL

A moveable panel houses all connection sockets to the EURORACK, except that to headphones (see drawings next to the front cover). The EURORACK comes supplied with this panel fixed to the rear of the unit, ready for flat-mounting the console. If you want to rack mount your EURORACK, you will probably want to re-locate the connector panel to a more convenient site on the underside of the chassis.

1) Lay the EURORACK on a soft surface, face down. Take care not to apply excessive pressure to the console during the following procedures. You will need a cross-head screwdriver to effect this manoeuvre.
2) A blank panel on the underside of the chassis covers the recess where the connector panel is to be re-located. Remove the screws which are fixing this blank panel to the sides of the EURORACK chassis.
3) Next remove the screws fixing the cover panel to the base of the EURORACK.
4) Turn your attention to the Rear Panel, where the connector panel is currently sited. Remove the screws that fix this connector panel to the upper edge of the EURORACK.
5) Now loosen (do not yet remove!) the screws fixing this connector panel to the to the sides or the EURORACK chassis. While you support the weight of the connector panel with one hand, remove the loosened screws with the other.
6) Carefully manoeuvre the connector panel into its new location, taking care not to snag or strain any of the ribbon cables that connect it to the rest of the console's circuitry. The bottom edge of the connector panel should now rest on a flange at the base of the chassis.
7) Loosely fit the screws that will fix the connector panel to the EURORACK's sides.
8) Now fit the cover plate into its new location on the back of the EURORACK. The folded edge of the cover plate always faces the right-angled edge between the base and back of the EURORACK.
9) Loosely fit the screws that fix the cover plate to the rear of the chassis, and the ones that complete the mounting of the connector panel on the underside.
10) Once you are satisfied that both panels are properly seated and all screws are on the correct locations, tighten up all screws. Do not force any screw. Check that you have aligned the panels correctly if any seem too tight.
11) The entire procedure may be reversed at any time.

8.2 CONNECTORS

You will need a lot of cables for different purposes - see fig. 8.1 to make sure you got the right ones.

All Outputs (except Direct Outputs and Insert Point Sends) are Ground-compensated (decoupled from the mains supply earth) to eliminate the possibility of ground loops.

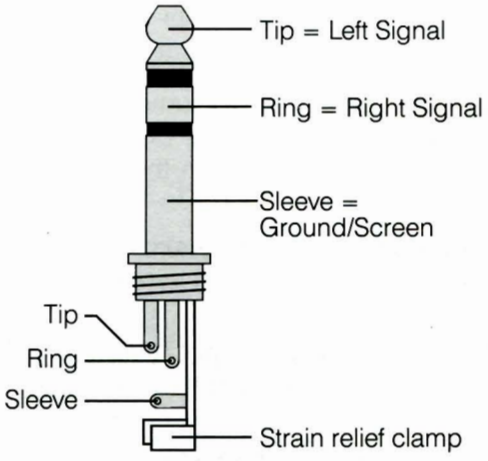
Unbalanced equipment may be connected to balanced inputs/out puts. Either use Mono 1/4" jacks or connect ring - barrel of TRS jacks (or pin 1 and 3 of XLR plugs).

Phantom power (+48 V D.C.) is provided. This can be switched on or off by the +48 V Phantom switch (S36).

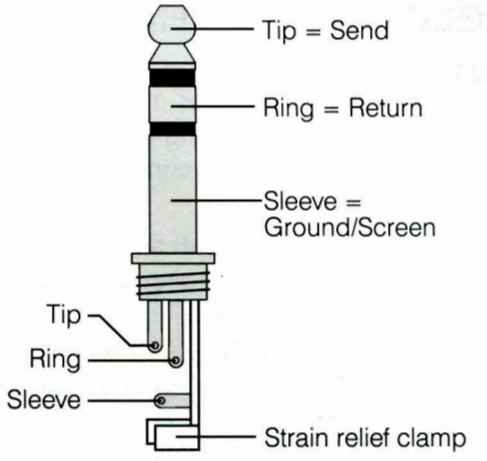


Care should be taken NOT to plug mics into the console (or stagebox) while the phantom power is on. Also, mute the monitor/PA speakers when turning phantom power on or off. Allow 1 minute after powering up for the system to equilibrate before setting input gains.

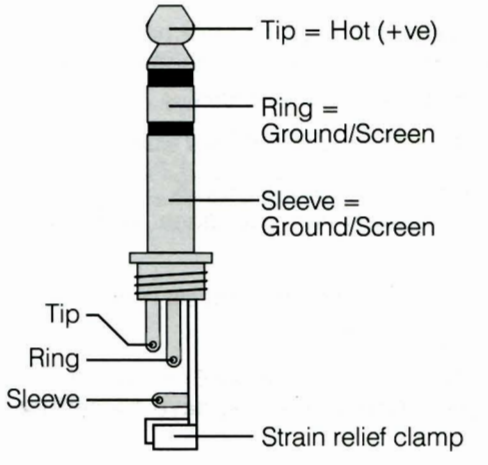
1/4" Stereo (TRS) Plugs



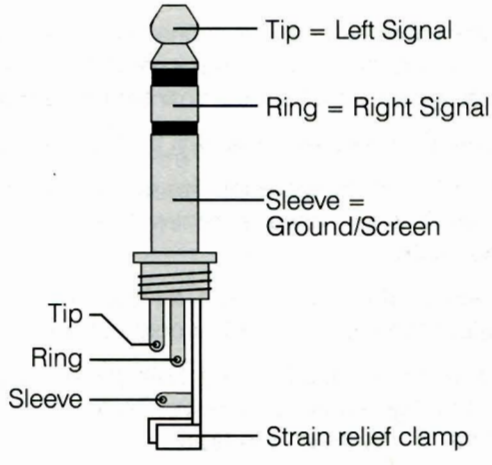
You'll need this kind for:
Headphones (stereo)



You'll need this kind for:
Inserts (unbalanced)



You'll need this kind for:
Direct Outs (unbalanced)



You'll need this kind for:
All Inputs
Mix Outputs
Aux Outputs
Group Outputs (MX 2642)
(balanced)

Male XLR plugs ('soldering' view of the cable connector)

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



You'll need this kind for balanced Mic input

For unbalanced use connect pin 3 to pin 1

Fig. 8.1: The right cable connectors for your EURORACK

9.0 PATCHFIELD

9

Nomenclature clarification:

FIELD = entire patching area, BAY = 1U 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 I 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. Plus, in a rack you can't get to the rear connections of individual units very easily!

9.1 THE NORMALIZED BAY

Most decent jackbays offer two rows of up to 24 normalized jacks in a 1u rack space. The term 'Normalized' 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 normalized insert pair does not break the internal connection, but it does provide an alternative direct output.

Where normalizing 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 Manufacturer's instructions for how to do this.

9.2 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 patch cords 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 levels (+4 dBu or -10 dBV). In any case, patching in a field with 48 V DC flying about is to be avoided at all costs! It is best to plug mics directly into the EURORACK, or via special XLR-type wall boxes connected to the EURORACK's mic inputs by a good quality balanced (2-core + screen) multicore cable (see also section 13.4: ((Un)balanced Lines).

MX 2642 Condensed Patchfield Example

for MX 2642 and 8-Track Recorder, incorporating Behringer ULTRALINK 662

Notes:

1) When using one of the MIC inputs (Channels 1-8) disconnect any jack plugged into that channel's Line Input to minimize noise and remove the possibility of multiple signal inputs to the channel.

2) In the configuration below, the ULTRALINK's inputs are normalled to the 4 Subgroup outputs, two mono and one stereo effects units, and the output is normalled to the mixer's 2-TK input. This means that you can apply dynamics processors to Subgroups, pan mono Subgroups, and have 3 (or 2 stereo) effects units all coming back on the 2-TK input, freeing up the remaining 24 inputs of the MX 2642 for tape and instrument tracks, mic channels and extra FX returns.

3) ▪▪▪▪▪▪▪▪▪▪▪▪▪▪ = Break the normalizing on this bay.

4) 'Tie Lines': usually, in a MIDI set-up, 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 strategically positioned near to where MIDI hardware congregates.

5) It is always useful 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 three different instruments on it. Patch the Direct out of its Channel into a 4-way split, returning to a further 2 channels via Line input. Set up one Channel for each instrument, and use mutes (preferably MIDI controlled) to mute the two unwanted Channels at any one time.

6) If you've got more than 4 or 5 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 or even insert points, and therefore require as many as 3 or 4 holes per channel.

Enhancers are usually applied across Inserts like compression and EQ etc., but Behringer's ULTRAFEX II has a Solo mode, in which it can be addressed via an Aux Send and blended back into the main mix like any reverb etc.

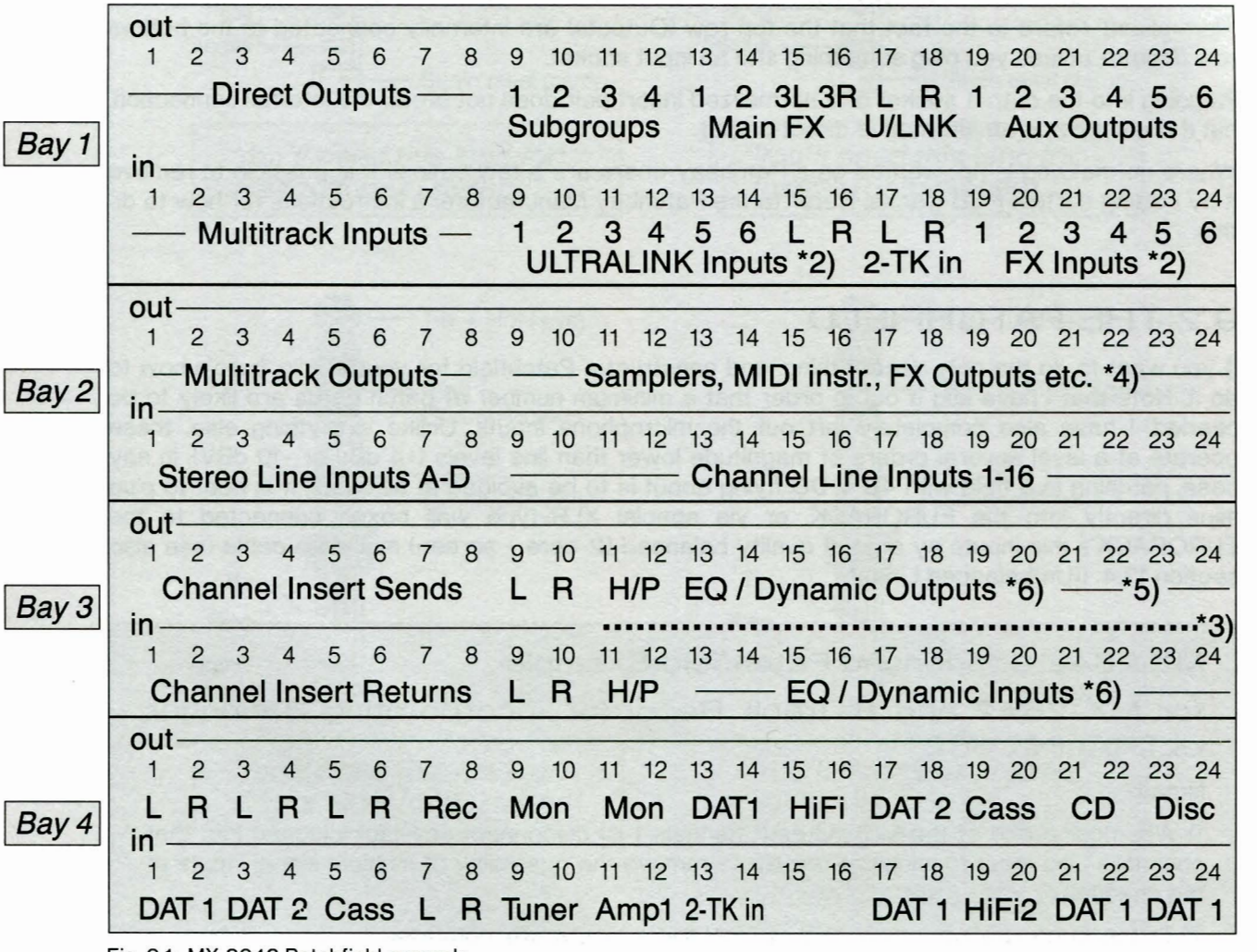


Fig. 9.1: MX 2642 Patchfield example

MX 2804 Condensed Patchfield Example

for MX 2804 and 8-Track Recorder

Notes:

1) When using one of the MIC inputs (Channels 1-8) disconnect any jack plugged into that channel's Line Input to minimize noise and remove the possibility of multiple signal inputs to the channel.

2) In the configuration below, the Channel Direct Outputs are normalled to tape tracks. When you want to lay more than one track to tape, patch the L/R Output and/or Aux Send(s) to tape track(s). When using the L/R Output, be sure to mute all Channels not involved.

3) ▪▪▪▪▪▪▪▪▪▪▪▪▪▪ = Break the normalizing on this bay.

4) 'Tie Lines': usually, in a MIDI set-up, 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 strategically positioned near to where MIDI hardware congregates.

5) It is always useful to have a few 4-way links around for splitting signals up to 3 ways (one in, three out). EG. One tape track has three different instruments on it. Patch the Direct out of its Channel into a 4-way split, returning to a further 2 channels via Line input. Set up one Channel for each instrument, and use mutes to mute the two unwanted Channels at any one time.

6) If you've got more than 4 or 5 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 or even insert points, and therefore require as many as 3 or 4 holes per channel.

Enhancers are usually applied across Inserts like compression and EQ etc., but Behringer's ULTRAFEX II has a Solo mode, in which it can be addressed via an Aux Send and blended back into the main mix like any reverb etc.

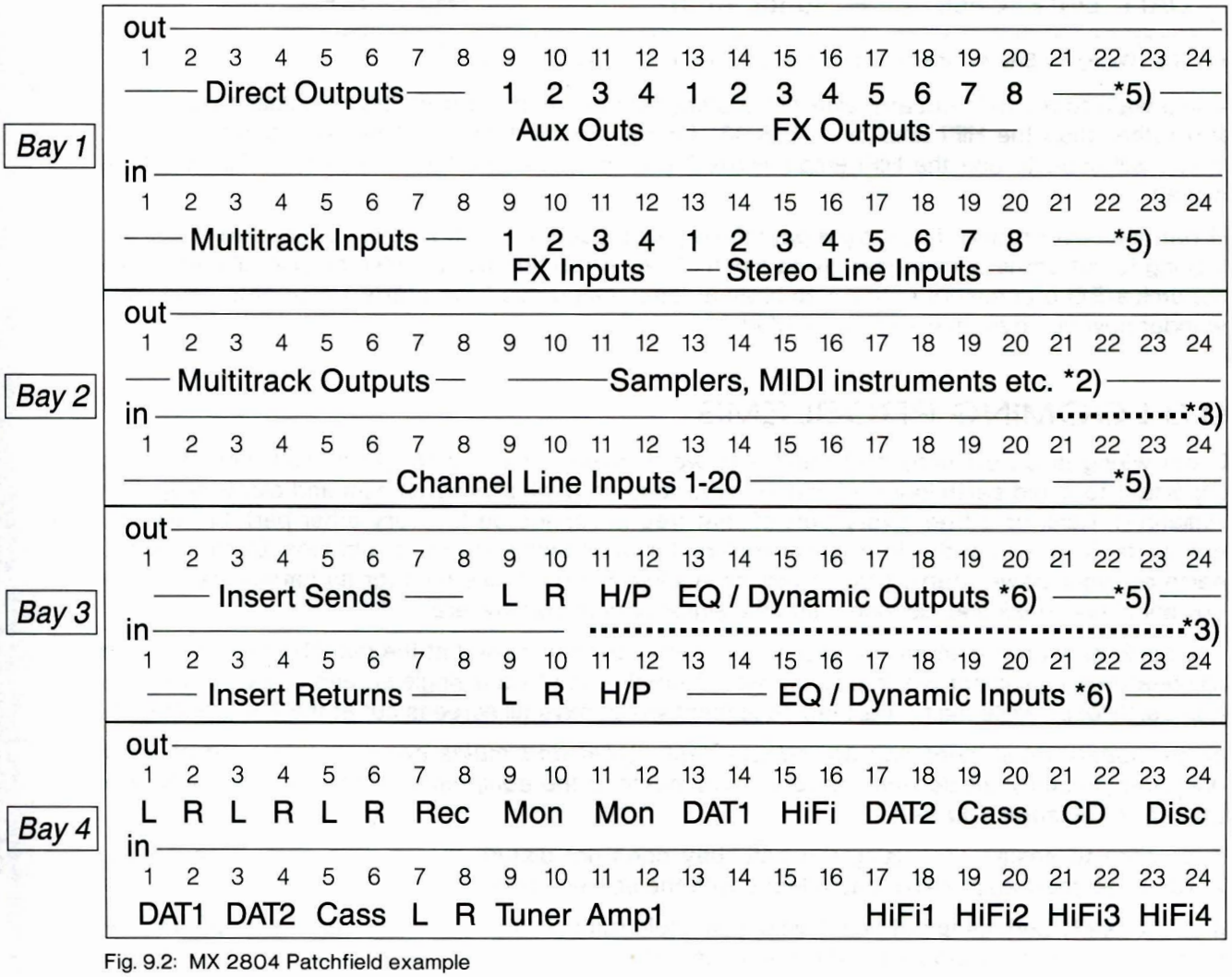


Fig. 9.2: MX 2804 Patchfield example

Bay 4 in Detail

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

Outputs 7 and 8 feed the L/R Rec input. I have hard-wired this to Rec outputs 1-6 in order to be able to send to all recorders simultaneously. Not only does this mean that you can simultaneously record a mix to all recorders. Copying from ANY 2-Track source to ALL recorders may now be done by patching the Source outputs into the L/R Rec Input.

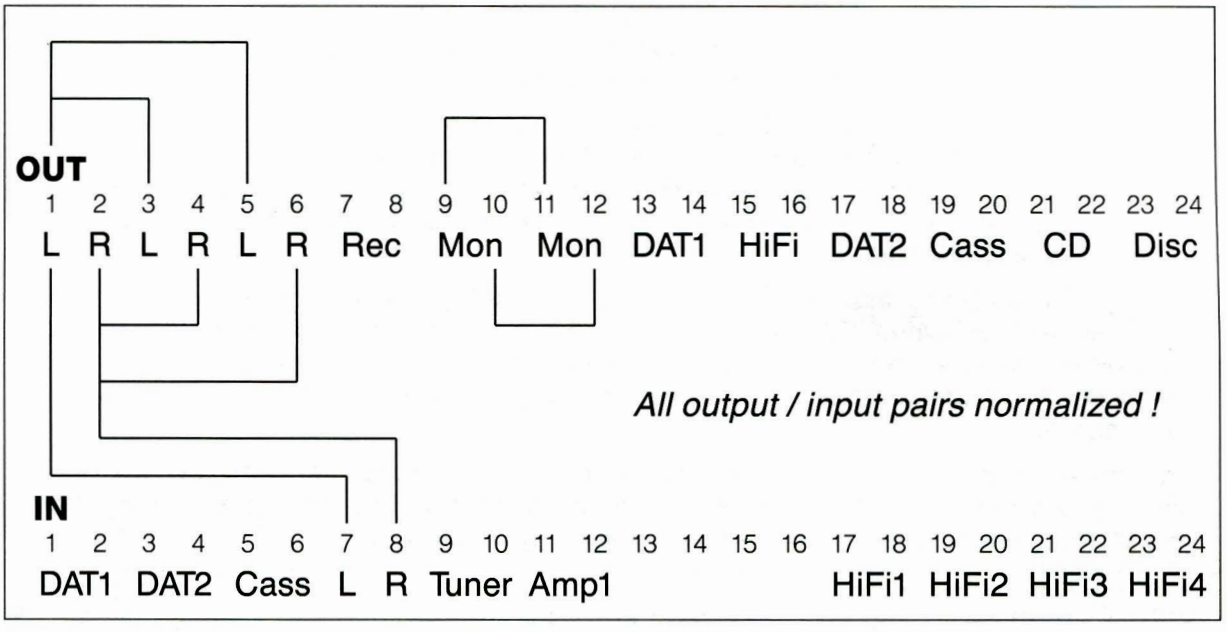


Fig. 9.3: Wiring for Bay 4 (for advanced wiring scheme refer to section 13.3)

If you want to record from any external source, best patch direct from the individual outputs (17 - 22) rather than the HIFI amp mix (15 & 16) for the cleanest result. (The exception being vinyl, which will need to use the HIFI amp's RIAA Pre-amp to present the mixer with a flat response signal).

I'll bet you never realized how big a patchfield you needed for such a compact console! If you are looking to cut corners, I suggest giving much of bay 4 top priority, and also placing all inserts and dynamics/EQ processors onto a patchbay at least. Once you have started your patchfield you'll wonder how you ever managed without it!

9.3 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 an ariel, picking up hum and 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 its 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 13.4: (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!

10.0 EXPANDING

10

When the EURORACK is your main mixer, you may find that you run out of inputs as your system expands. It is possible to expand your mixing system by combining two or more mixers.

Adding extra Line Inputs to your EURORACK

A small line mixer (such as the Behringer ULTRALINK MX 662 6+2 Channel Mixer/Splitter/Signal Router) can inexpensively add extra line inputs to your console. With the ULTRALINK, any stereo, line-level Input on your EURORACK can become a Stereo Line Input plus a further 6 pan-able Mono Line Inputs. Great for adding Tape Monitor Returns etc.

Linking two EURORACKS

Simply take the Main Mix Output of one, and feed it into a suitable Stereo Line-level Input of the second. The signal from EURORACK 1 is a Submix of up to 26/28 channels of sound, and is therefore likely to be considerably higher than the typical single source signals (coming from mics, MIDI instruments or tape) seen by the remaining channels of EURORACK 2. Reduce input gain on the EURORACK 2 channels which are accepting the output from EURORACK 1 (according to the Setting Up Procedure: see Section 11). The final fader settings on EURORACK 2 should show about a 10 dB average discrepancy between the return from EURORACK 1 and all other Inputs.

Linking the EURORACK to a Master Console

In order to integrate your EURORACK fully with an existing console's architecture we suggest you make the following connections. All Outputs (L, R, Subgroups (MX 2642 only) and Aux Sends) should be patched into individual channels or line inputs of the main console. In particular each EURORACK Aux Send should be routed (probably via a channel on the main console) into a separate aux send bus of the main console, if you want the EURORACK to send to effects already in use by the main console.

Another way to link the two consoles would be to simply feed the stereo output of one into a suitable stereo line (or two mono line) inputs of the other.



The EURORACK Outputs are essentially submixes of several channels of sound, and are therefore likely to be considerably higher than the typical source signals (coming from mics, MIDI instruments or tape) seen by the remaining channels of the master console.

11.0 SETTING UP

11

11.1 SELECTING INPUTS

1) Mono Channels accept mic or line inputs. If you are using the mic input, make sure nothing is connected to the line input (and vice-versa).

The mic inputs are more sensitive than the line inputs. Do not connect mics with Phantom Power switched on. NEVER use unbalanced mic cables with the Phantom Power switched on EVER!!!! Shorting 48 V to earth can cause serious damage.

2) Stereo Channels accept -10 dBV or +4 dBu line level signals. Any stereo channel can be run in mono simply by connecting into the left jack socket only. These channels are suitable for a variety of line-level sources including MIDI instruments and Tape Returns from multitrack.

3) Stereo Line Inputs are primarily designed for returning effects units, though these too may be given over to multitrack returns or MIDI instrument outputs.

11.2 INITIALIZING CHANNELS FOR GAIN-SETTING

1) Set gain to minimum and all Aux Sends to OFF (fully counterclockwise).

2) Set EQ to flat (all knobs at 12 o'clock).

3) Where applicable, set Lo Cut switch (S2) ON for most mics, OFF for signals with desired very low frequency content.

4) Set Channel Mode to PFL (S37 up).

5) Depress SOLO/PFL (S14) switch.

11.3 AUDITIONING A SIGNAL AND SETTING UP A CHANNEL

1) Make a typical noise, or roll the tape. There should also be some activity at the main L/R bargraph meters, indicating the PFL level.

2) For Mic/Line Inputs: Adjust the Gain control (P1) until transient peaks are regularly hitting +6 dB. Continuous signals should not exceed 0 dB.

3) For Stereo Line Inputs: Adjust the Gain control (P22) until transient peaks are regularly hitting +6 dB. Continuous signals should not exceed 0 dB.

4) If EQ is used, repeat steps 11.2 1) & 2)

5) If an Insert is used to patch in a compressor, gate, EQ etc, use the outboard processor's Bypass or Effect Off switch to A/B monitor the effect. If it does not have a bypass switch or equivalent, you will have to keep connecting and disconnecting the device until you complete the following procedure: Adjust the processor's output level so that effected and bypassed signals are of comparable level, ie 'unity' gain.

6) SOLO/PFL switch (S14) UP. Move onto next channel.

11.4 DESK NORMALIZATION

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 counterclockwise etc. Many controls have a natural initial setting. For EQ cut and boost this is centre position. However, some settings, such as selecting Pre or Post for channel Aux sends, will depend on the operating environment (e.g. Studio or Live), or on a particular engineer's preferred way of working.

11.5 MULTITRACK INITIALIZATION

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 optimized before recording commences.

11.6 RECORDING LEVELS

When recording to digital, it's a good idea to keep the recorder's peak meters below 0 dB. Most (not all, esp. 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. If you REALLY want to take your recording level to the limit (and fully exploit 16-bit digital's 96 dB dynamic range), you'll have to do some calibrating. How to do it? Well, 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 be way under its maximum input limit. Probably a better way to work out just how hard you can drive your recorder is to incrementally increase the record level until the onset of digital distortion, subtract, say, 5 or 10 dB, and never exceed that level. Engage 'peak hold' on your recorder before recording if you want to confirm that you haven't.



The Behringer PROTECTOR 8-channel Recording Limiter is specially designed to prevent digital distortion. It's extremely easy to use - a real set and forget device - and it not only enables you to use digital's full dynamic range, but it also can increase average digital recording levels, something previously done by saturating analogue tape.

When recording to analogue, the tape machine's VU meters should show around +3 dB 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. Hi-hats should read about -10 dB on a VU meter, as against 0 dB for a typical snare drum, and +3 dB or more for a kick drum.



Peak meters read more-or-less independent of frequency. Aim for 0 dB recording level for all signals.

MX 2642 only

Lining Up Recorder/Sampler Inputs

Set the channel sensitivity to match the operating level of your multitrack (consult the manual, phone the manufacturer, or simply 'suck & see' to find which setting works best). The sampler's variable/switchable input gain range is bound to accept -10 dBV and/or +4 dBu. There is no oscillator in the EURORACK, but you can use a simple unmodulated sustained tone from a keyboard. Choose one around 1 kHz. Set the channel EQ to normal (12-o-clock), and line up the channel according to the SETTING UP procedures (11.2). Route this signal to all Subgroups and adjust the Subgroup Output faders so that 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 0 dB on the tape recorder's input meters. A discrepancy of 12 dB indicates that a wrong operating level has been selected. Now adjust the sampler's input level until it also reads 0 dB.



Beware of inaccurate/uncalibrated sampler input meters. Work out how hard you can safely drive the sampler's input, reference this to 0 dB on a EURORACK group (PFL), then take note of the sampler's input gain pot setting. (Or use soft adhesive tape etc. to hold it in one position). Now you can use the Subgroup meters to confirm level when sampling, usually far clearer than using the Sampler's own input level display.

MX 2642 only

11.7 AUDITIONING A MIX

In order to be heard other than when Solo/PFL-ed, a channel must be routed to the Main Mix bus. This can be either from the Channel routing matrix directly (S20), or via one of the Subgroups (S18/19), so long as that Subgroup is itself routed to the L/R bus (S35). Channels going to tape are usually monitored via Tape Returns. On the EURORACK this means using either Input Channels (stereo or mono), Stereo Line Inputs, or via a separate submixer (e.g. Behringer ULTRALINK MX 662), patched into a suitable input such as that for 2-Track playback. The Main Mix button of Channels and Subgroups going to Tape should be UP, so as to avoid 'double' auditioning.

11.8 TRACK SHEET

When laying out channels for recording or mixing, try to be sensible. Keep tom-toms together, etc. Work out a scheme that suits you and stick to it. A common order is: kick drum, snare, hi-hat, tom-toms (as the audience sees the kit), cymbals (ditto), bass, guitars, keyboards, other instruments, vocals. From session to session and gig to gig you will soon know where you are without hardly ever having to look at a tracksheet.

12.0 APPLICATIONS

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12.1 KEYBOARD MIXING, LIVE OR IN THE STUDIO

This is relatively simple to achieve. Simply use the line inputs to mix stereo or mono outputs from your keyboards. *For MX 2642 users,* subgrouping may be useful en route to the mix, e.g. to control the level of drums versus music. Aux Sends may be used either to feed on-stage monitors, artists headphones or effects units. Use the Stereo Line Inputs A-D for instruments with built-in EQ, since there is no desk EQ on these inputs. The Main Mix Output should feed the FOH or studio mixer. Two typical comprehensive live set-ups are shown below:

MX 2642 Example 'Keyboard mixing'

Source	*Route*	*Mic/ Line*	*Mono/ Stereo*	*Input*
MIDI instrument #1	1/2	L	M	1
MIDI instrument #2	1/2	L	M	2
MIDI instrument #3	1/2	L	M	3
MIDI instrument #4	1/2	L	M	4
MIDI instrument #5	1/2	L	M	5
MIDI instrument #6	1/2	L	M	6
MIDI instrument #7	1/2	L	M	7
MIDI instrument #8	1/2	L	M	8
MIDI sampler #1	Mix	L	S	9/10
MIDI sampler #2	Mix	L	S	11/12
MIDI drum machine #1	3/4	L	S	13/14
MIDI drum machine #2	3/4	L	S	15/16
Effects unit #1			S	A1/A2
Effects unit #2			S	B1/B2
Effects unit #3			S	C1/C2
Effects unit #4			S	D1/D2
Backing music			S	2-TK IN

Subgroup Out 1/2	› ASSIGNED TO MAIN MIX
Subgroup Out 3/4	› ASSIGNED TO MAIN MIX
Aux Out 1	› EFFECT (POST-FADER)
Aux Out 2	› EFFECT (POST-FADER)
Aux Out 3	› EFFECT
Aux Out 4	› EFFECT
Aux Out 5	› NOT USED
Aux Out 6	› NOT USED
Mix Out L	› COMPRESSOR (INSERT) › FOH
Mix Out R	› COMPRESSOR (INSERT) › FOH
Monitor Out L	› PERSONAL FOLDBACK (MAIN STEREO FEED)
Monitor Out R	› PERSONAL FOLDBACK (MAIN STEREO FEED)

MX 2804 Example 'Keyboard mixing'

Source	*Mic/ Line*	*Mono/ Stereo*	*Input*
MIDI instrument #1	L	M	1
MIDI instrument #2	L	M	2
MIDI instrument #3	L	M	3
MIDI instrument #4	L	M	4
MIDI instrument #5	L	M	5
MIDI instrument #6	L	M	6
MIDI instrument #7	L	M	7
MIDI instrument #8	L	M	8
MIDI sampler #1	L	S	9/10
MIDI sampler #2	L	S	11/12
MIDI synth #1	L	S	13/14
MIDI synth #2	L	S	15/16
MIDI drum machine #1	L	S	17/18
MIDI drum machine #2	L	S	19/20
Effects unit #1		S	A1/A2
Effects unit #2		S	B1/B2
Effects unit #3		S	C1/C2
Effects unit #4		S	D1/D2

Aux Out 1	› EFFECT (POST-FADER)
Aux Out 2	› EFFECT (POST-FADER)
Aux Out 3	› EFFECT
Aux Out 4	› EFFECT
Mix Out L	› COMPRESSOR (INSERT) › FOH
Mix Out R	› COMPRESSOR (INSERT) › FOH
Monitor Out L	› PERSONAL FOLDBACK (MAIN STEREO FEED)
Monitor Out R	› PERSONAL FOLDBACK (MAIN STEREO FEED)

12.2 LIVE GIG WITH SIMULTANEOUS 8-TK RECORDING

Here some or all mono channels are likely to be tied up with stage mics. Carefully position these so as to minimize feedback. Try to keep the stage volume as low as possible, as stage sound can interfere with and muddy FOH sound, as well as causing a reduction in feedback thresholds. Don't forget to notch out troublesome frequencies using a Graphic or Parametric Equalizer, or Feedback Killer (see the Behringer ULTRA-CURVE DSP 8000, which does all of these and more).

Use the Lo Cut Filters to eliminate floor rumble, mics popping etc.
Try using a compressor and noise gate on vocals, bass, guitar and even drums via Channel Insert.

MX 2642 Example 'Live Gig with 8-TK Recording'

Source	*Route / To Tape*	*Mic/ Line*	*Mono/ Stereo*	*Input*
Vocals	Mix / Direct 1	M	M	1
Backing Vocals	Mix / Direct 2	M	M	2
Kick Drum	Mix / 1/2	M	M	3
Snare Drum	Mix / 1/2	M	M	4
Drumkit L	Mix / 1/2	M	M	5
Drumkit R	Mix / 1/2	M	M	6
Bass	Mix / Direct 3	M or L	M	7
Guitar	Mix / Direct 4	M or L	M	8
Keyboard #1	Mix / 3/4	L	S	9/10
Keyboard #2	Mix / 3/4	L	S	11/12
Keyboard #3	Mix / 3/4	L	S	13/14
Keyboard #4	Mix / 3/4	L	S	15/16
Effects Unit #1			S	A1/A2
Effects unit #2			S	B1/B2
Effects unit #3			S	C1/C2
Effects unit #4			S	D1/D2
Intro Music			S	2-TK IN
Subgroup Out 1/2	› Route to Mix / Send to Tape Tracks 5/6			
Subgroup Out 3/4	› Route to Mix / Send to Tape Tracks 7/8			
Aux Out 1	› CUE 1 (PRE-FADER)			
Aux Out 2	› CUE 2 (PRE-FADER)			
Aux Out 3	› EFFECT			
Aux Out 4	› EFFECT			
Aux Out 5	› EFFECT			
Aux Out 6	› EFFECT			
MIX Out L	› GRAPHIC EQ (INSERT) › FOH			
MIX Out R	› GRAPHIC EQ (INSERT) › FOH			
Monitor Out L	› SIDEFILLS (MAIN STEREO FEED)			
Monitor Out R	› SIDEFILLS (MAIN STEREO FEED)			

With the MX 2804 being a two bus console, it is not possible to route to tape via Subgroups. Therefore Direct Outs and the Main Mix are used. All instruments and vocals will be recorded in the Main Mix which feeds the PA as well as two recording channels. The further 6 channels of the multitrack are used to carry the most important instruments and vocals, to enable a degree of blending during mixdown later. These can be raised in level relative to the overall mix that took place at the concert.

MX 2804 Example 'Live Gig with 8-TK Recording'

Source	*Route / To Tape*	*Mic/ Line*	*Mono/ Stereo*	*Input*
Vocals	Direct 1	M	M	1
Backing Vocals	Direct 2	M	M	2
Kick Drum	Direct 3	M	M	3
Snare Drum	Direct 4	M	M	4
Drumkit L	Mix	M	M	5
Drumkit R	Mix	M	M	6
Bass	Direct 7	M or L	M	7
Guitar	Direct 8	M or L	M	8
Keyboard #1	Mix	L	S	9/10
Keyboard #2	Mix	L	S	11/12
Keyboard #3	Mix	L	S	13/14
Keyboard #4	Mix	L	S	15/16
Turntable	Mix	L	S	17/18
Intro Tape	Mix	L	S	19/20
Effects Unit #1			S	A1/A2
Effects unit #2			S	B1/B2
Effects unit #3			S	C1/C2
Effects unit #4			S	D1/D2
Aux Out 1	› CUE 1 (PRE-FADER)			
Aux Out 2	› CUE 2 (PRE-FADER)			
Aux Out 3	› EFFECT			
Aux Out 4	› EFFECT			
MIX Out L	› GRAPHIC EQ (INSERT) › FOH			
MIX Out R	› GRAPHIC EQ (INSERT) › FOH			
Monitor Out L	› SIDEFILLS (MAIN STEREO FEED)			
Monitor Out R	› SIDEFILLS (MAIN STEREO FEED)			

12.3 8-TRACK MIDI PROJECT STUDIO

With largely computer-generated dance music you will want to have plenty of Line Inputs, and an ability to take vocals quickly, efficiently, and with minimal desk disturbance. Often a vocal line is added after the music is almost complete. For this we try NOT to use a valuable Aux Send as a Cue Feed. In general, the mix in the artist's cans can be the same as that going into the control room monitors: basically a stereo mix with 1) the vocal channel raised above 'mix' level in volume and 2) any off-putting channels muted.

As the vocal is sent DIRECT to multitrack, it will be recorded dry. If you intend to take several tracks in quick succession, why not feed the Direct Output simultaneously into, say, 4 tape tracks via a simple junction box, splitter lead, line mixer like our MX 662, or patchbay.

MX 2642 Example '8-Track MIDI Project Studio'

Source	*Route / To Tape*	*Mic/ Line*	*Mono/ Stereo*	*Input*
Vocals (Tape)	Mix / Direct	M (L)	M	1
Tape	Mix	L	M	2
Tape	Mix	L	M	3
Tape	Mix	L	M	4
Tape	Mix	L	M	5
Tape	Mix	L	M	6
Tape	Mix	L	M	7
Tape	Mix	L	M	8
Sampler #1	Mix	L	S	9/10
Sampler #2	Mix	L	S	11/12
Sampler #3	Mix	L	S	13/14
Sampler #4	Mix	L	S	15/16
Effects unit #1			S	A1/A2
Effects unit #2			S	B1/B2
Effects unit #3			S	C1/C2
Effects unit #4			S	D1/D2
DAT			S	2-TK IN

Subgroup Out 1/2	› AVAILABLE FOR MIXDOWN OR TRACK-LAYING
Subgroup Out 3/4	› AVAILABLE FOR MIXDOWN OR TRACK-LAYING
Aux Out 1	› EFFECT
Aux Out 2	› EFFECT
Aux Out 3	› EFFECT
Aux Out 4	› EFFECT
Aux Out 5	› EFFECT
Aux Out 6	› EFFECT
Mix Out L	› DAT
Mix Out R	› DAT
Monitor Out L	› CONTROL ROOM / HEADPHONES AMP
Monitor Out R	› CONTROL ROOM / HEADPHONES AMP

Note that there are 6 Aux Sends shown, but only 4 Stereo Aux Returns. Perhaps you have a couple of true Stereo processors (e.g. an ULTRAFEX in SOLO mode) requiring L and R feeds. Otherwise, you might like to use the 2-Track Input as an extra Stereo Aux Return. The smart move, however, might be to expand your console, either with the 8-channel ULTRALINK MX 662, or another EURORACK, using the 2-Track-to-Mix facility to blend the output of one mixer into the other.

MX 2804 Example '8-Track MIDI Project Studio'

Source	*Route / To Tape*	*Mic/ Line*	*Mono/ Stereo*	*Input*
Vocals (Tape)	Mix / Direct	M (L)	M	1
Tape	Mix	L	M	2
Tape	Mix	L	M	3
Tape	Mix	L	M	4
Tape	Mix	L	M	5
Tape	Mix	L	M	6
Tape	Mix	L	M	7
Tape	Mix	L	M	8
Sampler #1	Mix	L	S	9/10
Sampler #2	Mix	L	S	11/12
Sampler #3	Mix	L	S	13/14
Sampler #4	Mix	L	S	15/16
Effects unit #5	Mix	L	S	17/18
Effects unit #6	Mix	L	S	19/20
Effects unit #1			S	A1/A2
Effects unit #2			S	B1/B2
Effects unit #3			S	C1/C2
Effects unit #4			S	D1/D2
Aux Out 1	› EFFECT			
Aux Out 2	› EFFECT			
Aux Out 3	› EFFECT			
Aux Out 4	› EFFECT			
Mix Out L	› DAT			
Mix Out R	› DAT			
Monitor Out L	› CONTROL ROOM / HEADPHONES AMP			
Monitor Out R	› CONTROL ROOM / HEADPHONES AMP			

12.4 AUXLESS HEADPHONES MIX

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. It is also not unusual for a vocalist to be drafted in to add some colour to the mix at a late stage. The simplest solution here is to feed the main mix into the artist's cans. If you (or they) are uncomfortable working with the mix, try using a simple line mixer (like the Behringer ULTRALINK MX 662) to blend the Direct Output from the mic channel (possibly via a dedicated FX unit to keep the vocalist happy) with the Main Mix, the combined stereo signal then being fed into the vocalist's cans.

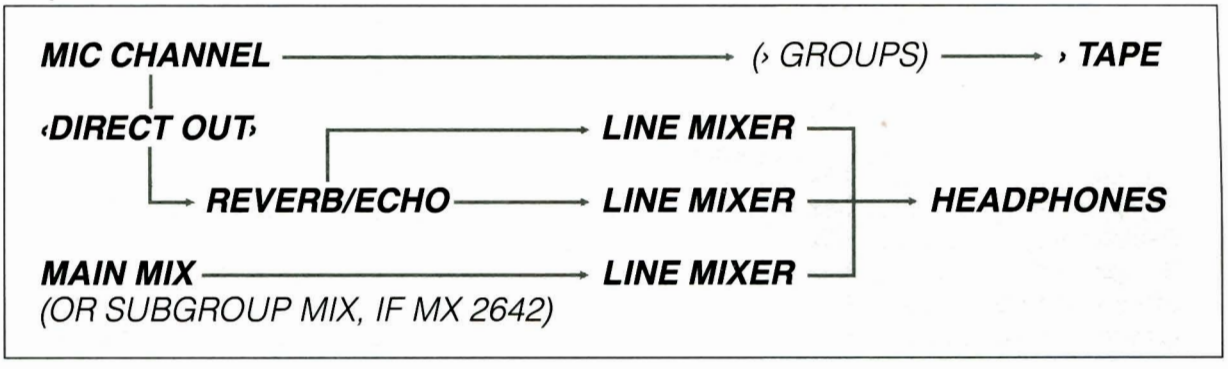


Fig. 12.1: Auxless headphone mix

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 are working with a MX 2642* and 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 Subgroup pair, omitting the offending channels, and use that output instead of the Main Mix to drive the headphones, either directly or via a line mixer as illustrated above.

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

13.0 SOME MIXING TOPICS

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13.1 EQUALIZATION

The variable parameters of the channel equalizers on the EURORACK are described in sections 3.2 and 4.2.

Few people buying a mixer will need to be told how an equalizer works. But how to get the best out of it? Well, 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 its 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.



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'. For sounds which require drastic corrective EQ, it is advisable to have a couple of channels of fully comprehensive parametric equalization in your rack. (You can always bounce tracks though the outboard EQ, freeing up the unit for the next task).

Check out the Behringer ULTRACURVE DSP 8000, a superlative digital Stereo Equalizer and much, much more. Or our ULTRA-Q PEQ 2000 5-band Constant-Q State-variable analogue EQ.

For 'Advanced Equalization', EQ might be applied to a signal as follows: First, trim the LF and HF shelves to achieve the required slope or 'loudness'. Now use a parametric EQ band to boost the most significant frequency for each instrument or tape track. Over all channels, if two or more of these frequencies coincide, then you might have to settle for second best in some cases, if you want to achieve optimum separation in the mix. *Really* nasty frequencies will need notching out.



A good vocal signal can be enhanced by applying a significant boost in the 12 kHz region or higher, above the nasty sibilance region. This is especially effective if you've got a de-esser patched Post-EQ.

Another expert tip: use the Lo 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).

With the LF set to boost, and the Lo Cut switch activated, you have pretty much got a Peak response rather than Shelving at the bottom. Good for tight but deep bass.



Remember EQ contouring can be done with Cut as well as with Boost. Cutting away the top and bottom, then pushing up the gain is equivalent to mid-range boost! EQ is not a one-way street!

Always re-set a channel's input gain (or external devices' output level) after altering the amount of desk EQ cut or boost applied.

13.2 GAIN OPTIMIZATION

PFL (Pre-Fader Listening) is *the* way to set a desk level. Setting up the channel input gain is discussed in the essential section 11. Master Aux Send levels are fixed at unity gain. 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 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. You'll have to rely on your ears to detect digital distortion, since different outboard processors calibrate their meters differently. If you hear distortion, but PFL-ing the Aux Return Input says you're just hitting 0 dB, then it must be coming from the Aux Send mixing amplifier or the FX unit itself. 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 Input.

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).

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.

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.

Analogue multitrack tape should be driven quite hard, since its 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. With digital tape or hard disk you have plenty of dynamic range, and treble pre-emphasis is not often necessary. Just don't let the signal distort!

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 !

13.3 ABOUT 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 up its own output!

In the Patchbay section I recommended that you parallel the EURORACK's output into all 2-TK 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 pretty 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.

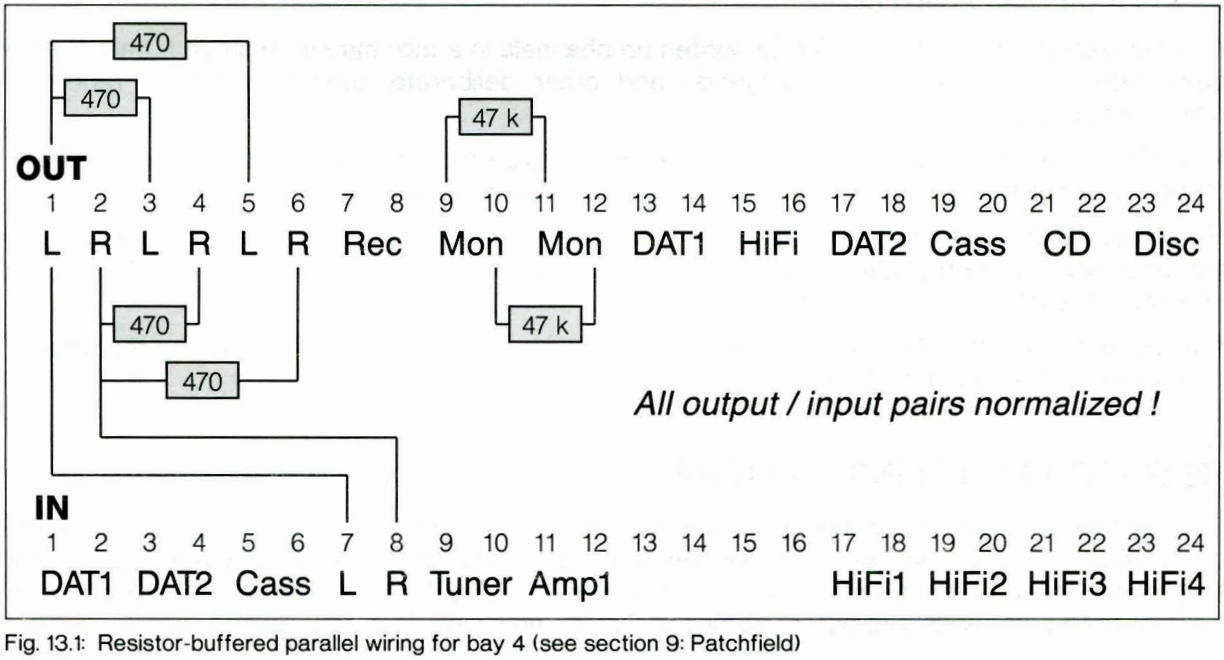


Fig. 13.1: Resistor-buffered parallel wiring for bay 4 (see section 9: Patchfield)

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 across one side of the monitor output un-buffered). Now, whenever you monitor an instrument's input level with the SOLO function, you can check it's tuning automatically. That should impress your customers. Especially those using old, unstable, but very en vogue analogue synths.

13.4 (UN)BALANCED LINES

Balanced inputs and outputs are offered on most audio connections on the EURORACK (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, 60, 100, 120 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 9.3: 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). You simply could not run a live P.A. rig without balanced mic lines. Although in the studio cable runs are shorter, the recorded medium's demands on signal-to-noise are far greater, and balanced mic lines are standard here, too.

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

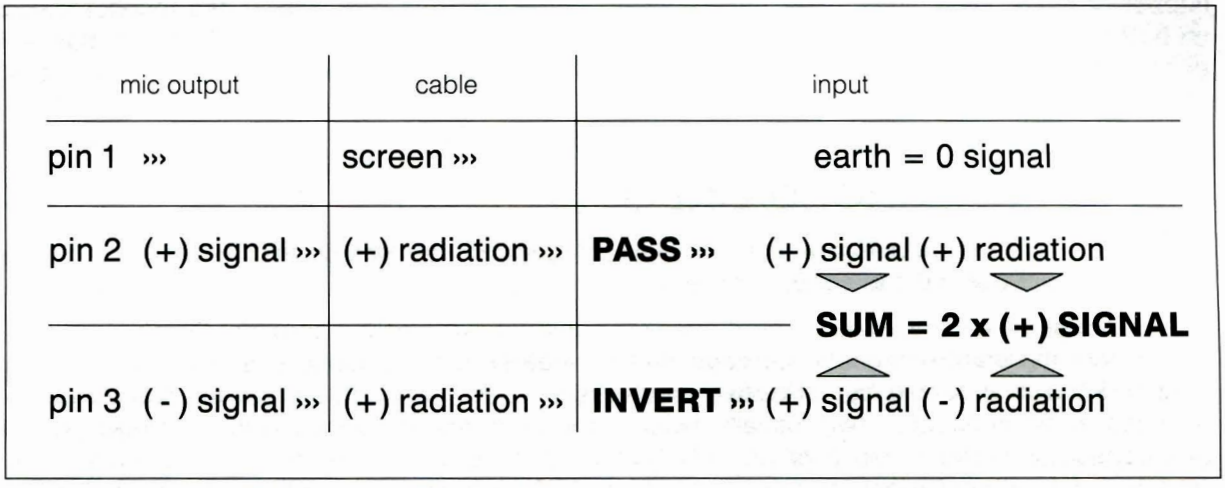


Fig. 13.2: A balanced mic line

14.0 RECORDING TOPICS

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14.1 TIMECODE

In analogue multitrack recording timecode usually goes down on the edge of the tape: track 8, 16 or 24, with the adjacent track left clear (Guard Band) to stop bleeding between recorded tracks and code. Digital formats *do not* waste any *audio* tracks on timecode: a separate sync is provided, usually at extra expense.

Ideally, timecode should be patched from the multitrack directly into the synchronizer's input. Normally a multitrack tape is *striped* with timecode before a session commences. Any sequenced music, click track, mix automation is then referenced to it. Always check sync before laying down any sequenced music: record a click track, then check to see that a 'live' playback doesn't drift. Timecode comes in various formats. The general rule is: format (and make/model/set-up of timecode generator) *must* be matched the reader.



If the gain from the recorder is too low to drive the sync unit,

1) *re-stripe at a higher level or*
2) *amplify the recorded timecode somehow, possibly via a desk channel not routed to any of the main buses. In this case use the channel's Direct Out to drive the synchronizer's input, in order to keep the timecode as remote as possible from the audio (timecode crosses over like nothing else I know).*

14.2 BOUNCING

Sometimes you want to play back from one or more tracks, route the signals to a new track or pair of tracks, and re-record. This process is commonly referred to as bouncing. Examples would be when reducing 4 lead vocal take tracks down to 1, reducing 4 separate tom-tom tracks down to a stereo pair via noise gates, putting a wild effect or EQ onto a dry signal, 'comping' backing vocal tracks, etc. etc.

On an analogue tape recorder, bouncing to an adjacent track is to be avoided. A feedback loop can be set up, since for all overdubs or bounces you will be using the same (sync) head for both playback and record. Organize your track layout so that any bounces you might have to do are always over *at least* one track.

There are no such restrictions when using digital multitracks.

14.3 SEQUENCING 'LIVE'

'Live' sequencing is a term used when a sequencer is driving MIDI instruments in a mixdown. Many remixers do not like to do this for several reasons.

1) It may be a problem assembling all the hardware/software used if a re-mix is required once the MIDI equipment has left the studio.
2) It is easier to send 1 or 2 24-track tapes around the world rather than have to hire in a truckload of gear for re-mixing.
3) MIDI timing problems mean that a tape master can sound tighter than a sequenced mix. Here sequencer tracks are laid one at a time, usually in sequencer Solo mode, to reduce MIDI congestion to a minimum.

Laying to linear tape has it's drawbacks. Apart from the sheer cost of hiring a 24/48 track studio, what happens if you want to change a song's structure half-way through?

Non-linear devices are still relatively expensive (though for how much longer is anyone's guess), and far the cheapest way to keep the arrangement flexible is to employ a sequencer. MIDI is sluggish (one event takes 0.66 milliseconds to pass down the serial MIDI bus), but you can get around this pretty well by following some basic rules.

Track Stack Rule
Sequencer tracks read from the top, therefore put the most timing specific tracks at the top. Start with rhythmic loops, then voices with fast transients, like percussion etc.

Midi Spread Rule
There are 2 ways to do this. Firstly, have as many MIDI ports as your computer will allow, and spread the MIDI activity as evenly as possible among them. Secondly, try to have as few MIDI notes as possible playing simultaneously, eg. advance slow attack tracks, put little silence onto the front of samples etc.

Who Benefits Most from Live Sequencing?
'Live' Sequence mixing is most suitable for enthusiasts who have all their own hardware permanently set up in one location. Re-setting a mix for correction is then relatively straightforward, especially if you can store instrument set-ups with your sequence data.

Sometimes re-mixes are performed away from the track-laying environment, or after the recording/mixing set-up (synths, hired out- board gear etc.) has been dismantled. For this you may have to have your music in a form that can be easily stored, shipped and recalled. To all intents and purposes, this means putting it onto a multitrack tape. However, the situation with dance music is much more flexible: many remixers only require a few elements of your composition to work with, and these can be shipped to a remote studio as DAT or proprietary sampler format samples.

15.0 SPECIFICATIONS

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Input Channels

Mic Input	Electronically balanced, discrete input configuration
Mic E.I.N. (22 Hz - 22 kHz)	129.0 dBu, 150 Ohm source 117.3 dBqp, 150 Ohm source 132.0 dBu, input shorted 122.0 dBqp, input shorted
Distortion (THD&N)	0.007% @ +4 dBu, 1 kHz, Bandwidth 80 kHz
Gain Range	+10 dB to +50 dB
Max Input (Mic)	+12 dBu
Line Input	Electronically balanced
Gain range	Unity to +40 dB
Max Input (Line)	+22 dBu
Channel Fader Range	+10 dB to -85 dB
Aux Send Gain Range	off to Unity to +15 dB

Equalization

Hi shelving	12 kHz +/- 15 dB, Q fixed at 2 oct.
Hi Mid shelving (MX 2642 only)	3 kHz +/- 15 dB, Q fixed at 2 oct.
Mid shelving (MX 2804 only)	2 kHz +/- 15 dB, Q fixed at 2 oct.
Lo Mid shelving (MX 2642 only)	500 Hz +/- 15 dB, Q fixed at 2 oct.
Mid sweep (MX 2642 only)	100 Hz - 8 kHz +/- 15 dB, Q fixed at 1 oct.
Lo shelving	80 Hz +/- 15 dB, Q fixed at 2 oct.
Lo Cut (High Pass) filter	75 Hz, 18 dB/oct.

Channel Direct Out

Max Output	+22 dBu
Noise @ Unity Gain	-94 dBu
Output Impedance	120 Ohm

Channel Insert

Max In / Out	+22 dBu
Channel to Channel Crosstalk	-95 dB @ 1 kHz

Subgroup Section (MX 2642 only)

Noise	Bus noise @ fader 0 dB: -102.5 dBr (ref.: +4 dBu); -92.5 dBr (ref.: +4 dBu, 16 channels assigned and set @ Unity Gain)
Submaster Output Max Out	+22 dBu balanced/unbalanced
Fader Range	+10 dB to -85 dB/off

Main Mix Section

Noise	Bus noise @ fader 0 dB: -101.0 dBr (ref.: +4 dBu); -92.5 dBr (ref.: +4 dBu, 16/20 channels assigned and set @ Unity Gain)
Max Output	+28 dBu balanced, +22 dBu unbalanced, 1/4" jacks
Aux Returns Gain Range	off to Unity to +20 dB
Aux Sends Max Out	+22 dBu

General

Distortion (THD&N)	0.007 %, @ +4 dBu, 1 kHz, Bandwidth 80 kHz; below 0.02 %, 22 Hz - 22 kHz at normal operating levels, any input to any output, Bandwidth 80 kHz
Frequency Response	20 Hz - 40 kHz +/- 1 dB any input to any output; 10 Hz - 120 kHz +/- 3 dB

Physical

Dimensions (H * W * D)	2.4"/6" (61/152.4 mm) * 17.15"/19" (435.6/482.6 mm) * 14" (355.6 mm)
Net weight	6 kg

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16.0 GLOSSARY

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Balance
relative levels of left/right in a stereo pair, usually controlled by a panoramic potentiometer (panpot)

Channel
input strip on a mixing desk

Clip, Clipping
overload, severe distortion

Cue
headphone feed

dB (decibel)
a unit of measurement, ratio of two voltages (dB = 20 log (V1 / V2)), (dBu = voltage ratio relative to 0.775 V RMS)

DI
connecting an electric instrument directly to the console via a DI-Box, converting the high level unbalanced signal (from the instrument) into a low level balanced signal (for the console)

Effects (abbr. FX)
devices that alter the original ('dry') sound or add something to it, e.g. delay

Equalizing
the use of filters for cutting or boosting of selected frequencies

Fader
linear potentiometer with logarithmic response

Feedback (howlround)
avoid it by not using a mic close to a speaker driven from the mic's own signal

Gain
degree of amplification

Group
mixer subgroup

Headroom
signal range between nominal level and clipping

Line level signals
from low impedance sources (-10 to +6 dBu)

Lo(w) Cut (= High Pass) Filter
cuts off low frequencies

Mixdown
process by which a multitrack recording is combined into e.g. one or two channels

Mute
(to engage) button for signal muting

Normal
connect an output to an input via breakable link

PFL
Pre-Fade(r) Listening

Phantom Power
+48 V DC voltage for powering condenser mics

Phase coherence
degree to which L & R signal are synchronized

Phase correlation
degree of mono compatibility of L & R signal

Pot or potentiometer
a variable, mostly rotary voltage driver used e.g. for gain or frequency control

Q
quality, bandwidth

Route
signal path (routing matrix = series of switches for route selection

Solo
solo-in-place, stereo image retained

Source
signal source (sourcing matrix = series of switches for source selection

Transient
a transitory (extreme) rise in signal level

Trimpot
variable gain potentiometer

Wet signal
mix signal with effects added

17.0 WARRANTY

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   Otto Brenner Str. 4
   FRG-47877 Willich/Munchheide II
   Federal Republic of Germany

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