When your reputation depends on your mixer you can depend on Dynamix.

The 3000 series is the result of several years of research and development. Not just at the test bench but talking to the people who count - people who depend on their mixer.

In order to meet your exact requirements, the Dynamix 3000 is fully modular. There are frame sizes to accommodate sixteen, twenty four and thirty two input channels. You can choose between output formats of 8:2 or 16:2. Flight cases are available for the state-of-art road user.

Each input is equipped with balanced XLR microphone inputs for low noise operation. 48 volt phantom powering, for studio condenser microphones, is switchable on every channel. Input sections are also equipped with phase reversal and a 20dB pad.

3000 series mixers feature extensive equalization. EQ sections are four band with 15dB cut or boost on all bands. Two overlapping mid-band controls are equipped with sweep frequency for pinpoint accuracy. Combined, these controls cover a range from 100Hz to 10kHz. The bass control is bell response, centred at 60Hz. The treble control is shelving response at 12kHz.

In order to meet todays foldback and outboard effects requirements, all Dynamix 3000 mixers have four auxiliary sends. Auxiliary 1 is pre-fade and normally used for monitor. The second aux control is assignable to either the aux 2 bus (pre-fade) or aux 3 bus (post-fade). Auxiliary 4 is post-fade and therefore most suitable as an effects send.

In addition, every input and output module has a separate send and return circuit with level controls. As a result, a 16:8:2 will accommodate 28 separate effects units with on board send and return levels for each. Simultaneously, the board will supply two foldback mixes - without inhibiting any of the normal input/output functions.

Routing buttons are conveniently located next to the input faders, which are available as ALPS. Fast and accurate metering is provided by 25 segment LED bargraphs. These are switchable for V.U. or P.P.I. response. All inputs are equipped with LED peak level indicators. Dynamix have equipped the 3000 series with comprehensive P.F.L. and A.F.L. monitoring on all inputs, outputs and auxiliary buses.

Subgroup and main outputs are on both ¼" jacks and XLR's. Balancing modules are available for broadcast use.

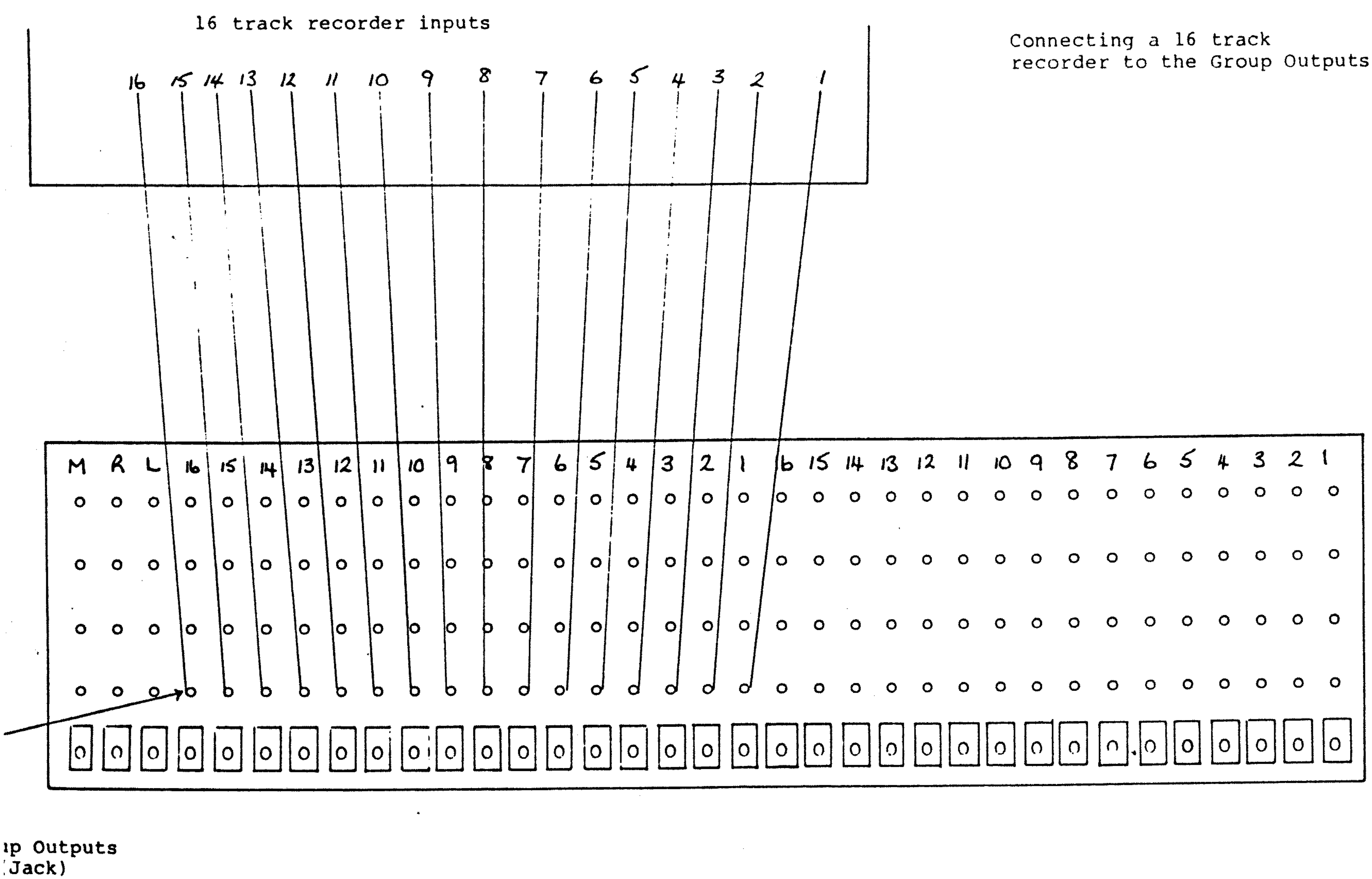
Our reputation depends on every mixer we make. That's why every 3000 we make is assembled by hand and rigorously tested before we ship.

Dynamix 3000 - You can depend on it.

DYNAMIX 3000 SERIES USERS MANUAL

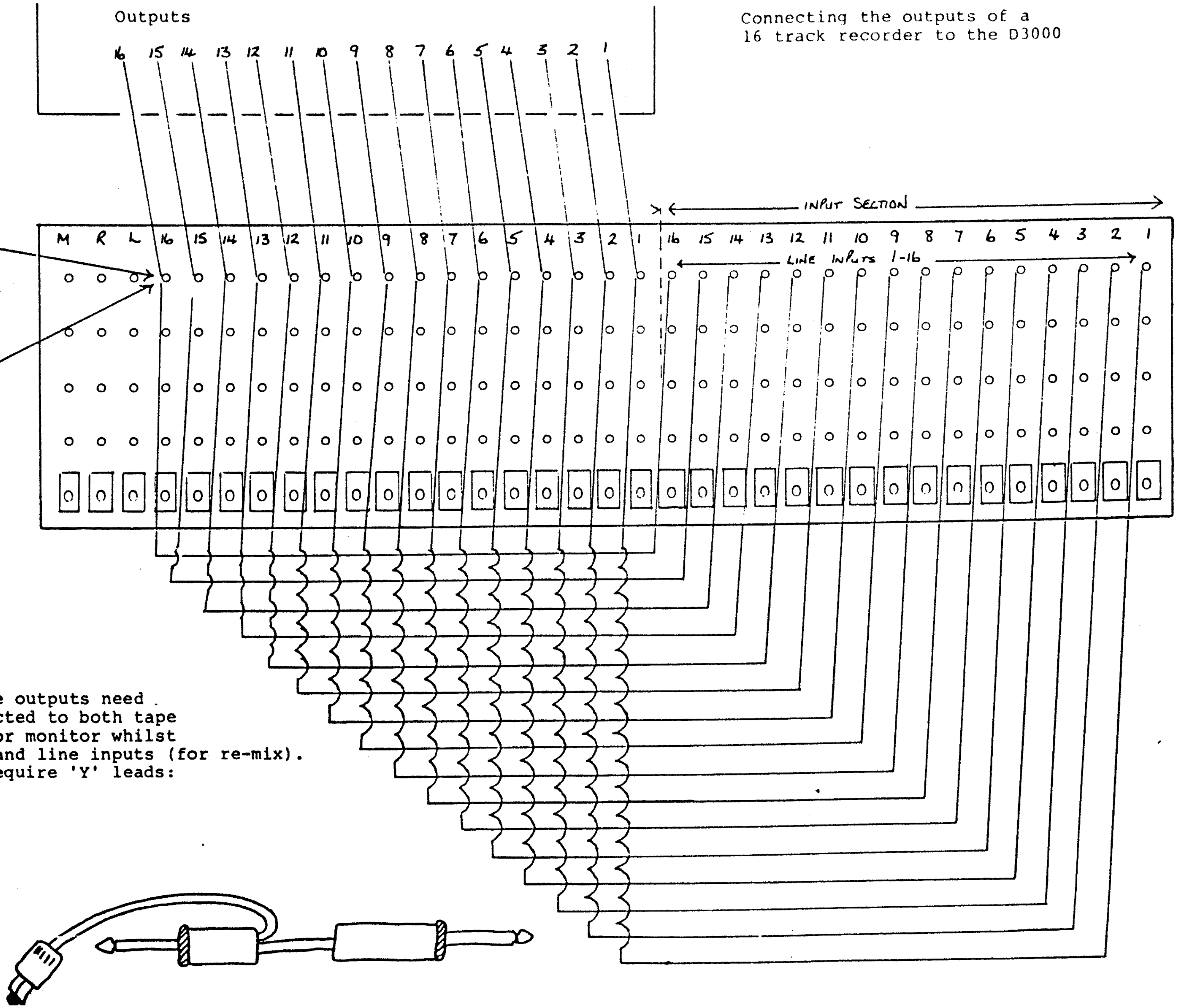
Contents:

Connecting a 16 track
recorder to the Group Outputs (Jack)



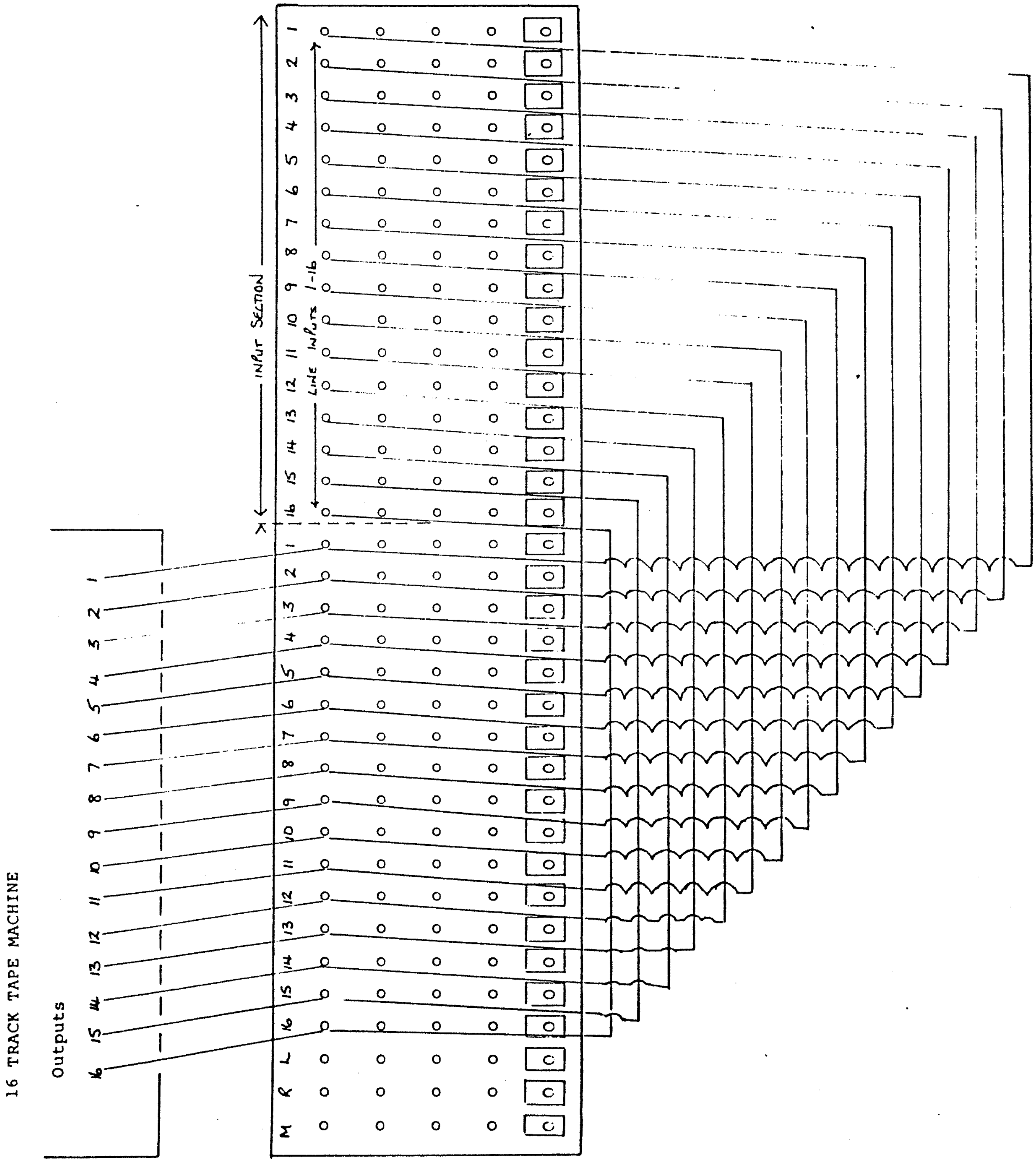
Connecting the outputs of a 16 track recorder to the D3000

16 TRACK TAPE MACHINE



Tape Monitors 1-16

Tape Machine outputs need to be connected to both tape monitors (for monitor whilst recording) and line inputs (for re-mix). This will require 'Y' leads:


16 TRACK TAPE MACHINE
Outputs
INPUT SECTION
LINE INPUTS 1-16
M
R
L

CONNECTING UP THE 3000

Earthing

Before making any other connections, please read the following section on earth connection.

CORRECT EARTHING IS ESSENTIAL FOR SAFETY AND AUDIO INTEGRITY

If your equipment is to operate correctly, it is important that it is earthed. Many of the other units in your set-up may already have individual earths connected to the mains plug. Unfortunately, when several units are connected in this way you will probably get an Earth Loop.

The only way around this problem is to earth all the equipment through one unit only. It is recommended that only the Dynamix mixer be connected direct to Earth. All other equipment will be earthed via the audio connections.

It is most important that you check the Earth on the mains supply that you intend to use. Simple checks can be made in two ways:

1) Physically examine mains sockets and power extensions to ensure that the earth wire is connected.

2) Use the illuminating type of test screwdriver to ensure that the Earth is not contaminated.

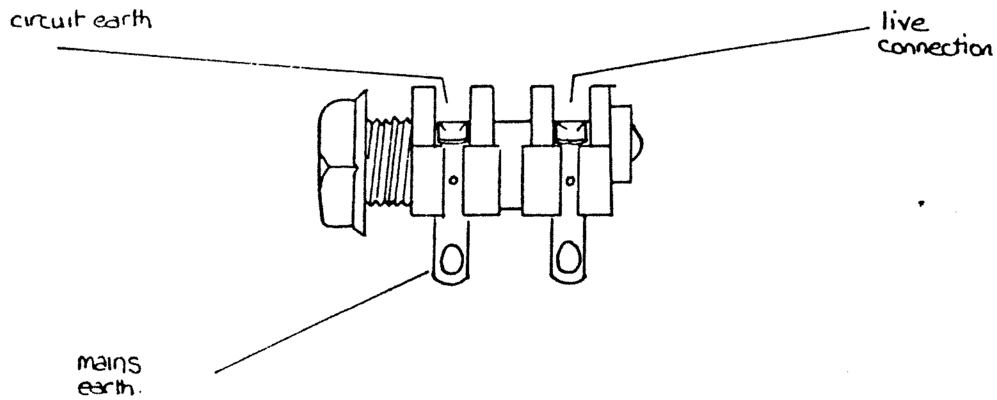
If you are in any doubt, or if there is not a proper earth connection, contact the Electricity Board.

When disconnecting the Earth wires from the other units be sure to secure the Earth so that there is no possibility of it touching a live connection.

Remember, no unit except the mixer will be earthed unless there is a continuous connection via the screen on the audio cable. If the screen is not connected both ends, the unit is not earthed.

There is a useful way of wiring the input jacks on the other equipment. Switching jack sockets will be needed. The circuit earth is connected to the sleeve terminal. This will earth through the mixer when a lead is inserted from the mixer. The mains earth is connected to the short sleeve terminal on the jack socket. When the jack plug is remove, circuit earth will automatically connect with the mains earth. The diagram will make this clearer. On no account attempt this conversion if you do not fully understand what is entailed.

Wiring for a Switching Jack Socket



The wiring above enables the mains earth to be disconnected when an audio lead is inserted into the jack

Some people prefer to earth all the equipment via the power amplifier. Dynamix advocate earthing through the mixer for the simple reason that this is the unit most often touched.

There is an argument in favour of using the power amplifier as earth. This is the unit that is likely to have the greatest earth ripple.

If you prefer to use the power amp for earth observe the following. The mains earth and the audio socket earth must be in direct contact. You can check this by metering the resistance between earth on the power lead and the amp chassis. It is suitable if there is a reading of zero OHMS.

Please seek qualified help if you do not feel confident about the wiring of your equipment.

AUDIO CONNECTIONS

Before the system can be used it will be necessary to make all the audio connections. A number of diagrams have been provided at the front of this chapter.

These show various ways in which a D3000 series mixer can be wired up. As the permutations available are almost endless, you may discover alternative wiring schemes that you prefer. Whether you are wiring up a permanent installation or a mobile P.A., cables are important. The Dynamix 3000 is capable of extremely high performance. It cannot however, function properly with defective cables or connectors.

All the pin configurations for the mixer sockets are shown at the back of this manual. Alternatively, ready made cables or wiring harnesses are available from professional audio dealers.

It is worth considering using multicore cable for the input connections. For mobile use, a multicore is essential. It not only reduces set-up time, it is also a good deal less obtrusive than 16 to 32 separate mic cables.

The 3000 series does not have a multipin connector fitted, for one thing, there is no agreed standard for multicore connectors. Secondly, connector failure is easier to locate and repair when the cable ends in separate XLR plugs.

However, the blank modules on a 3000 series have a rectangular cut out. This will accept an Edac or Varelco 56 contact socket.

A WORD ABOUT IMPEDANCE

"IMPEDANCE: Total opposition to an alternative current flowing through a circuit, equal to the square root of the sum of the squares of the resistance and the reactance"

Don't worry. You don't have to understand the above to work out impedance compatability.

Any piece of wire will create a resistance. The longer the wire, the greater the resistance. We can create a component with a measured resistance and this is called a resistor.

O.K. so far?

Impedance is not dissimilar to resistance. But where resistance is a D.C. measurement, impedance is an A.C. measurement.

Impedance is shown as X amount of OHMS and is depicted by the sign 'Ω'. (And you thought all microphones were Leo's, huh?)

MICROPHONE IMPEDANCES

Microphones typically come into two impedance categories: Low and High. Low impedance microphones are normally between 200 and 600 OHMS. High impedance microphones have an impedance in the region of 47K to 50K OHMS.

Low impedance microphones are technically superior to high impedance, after about six meters of cable run, high impedance mics start to lose high frequency response.

High impedance microphones were once popular, for the simple reason that it was expensive to make a low impedance valve amplifier. For this reason almost all guitar pickups are high impedance.

Your mixer is designed to accommodate low impedance microphones.

MICROPHONE INPUTS ON THE MIXER

Whilst the mixer is nominally designed for a 200 OHM microphone, it will work equally well with a 600 OHM microphone.

This is because the input impedance of the mixer itself is not 200 OHM. In order to work properly, the input impedance of the mixer has to be at least 10 times higher than the impedance of the microphone.

A 600 OHM microphone will need an input impedance of at least 6K OHM. The input impedance of your mixer is high enough to suit all low impedance microphones.

MAKING A MULTITRACK RECORDING

Contents:

This chapter will provide you with the basic procedures to complete a multitrack session. As the techniques are virtually identical, eight and sixteen track recordings are dealt with in the same chapter.

Dynamix 3000 mixers offer a wealth of possibilities. There are often several ways of accomplishing the same result. Rather than produce a mind-boggling comprehensive chapter, we will explain one way of recording.

Once you are familiar with the functions of the 3000 you may want to explore some further options. The second chapter covers more advanced techniques. Often, particular techniques are selected for artistic considerations. The mixer functions will soon become second nature, allowing you to concentrate on the creative side of recording.

On its own, this manual will make hard reading. Use it with the mixer and everything will make a great deal more sense.

PLANNING THE SESSION

For the recording session to run smoothly, a little pre-planning will be useful and save a lot of frustration later on.

It is particularly important for commercial facilities to ensure that all the necessary materials are to hand. These will include:

- Inter-connection and patching leads
- Tape for the multitrack and stereo machine
- Head cleaner
- Track planners
- All microphones, DI boxes etc.

A track planner is extremely useful. It enables you to see at a glance what is recorded on each track.

If a definite arrangement has been decided upon, it will be possible to work out the track allocation on paper.

It is worth bearing in mind that the outer tracks on a tape are always more prone to high frequency loss. These are best saved for bass instruments.

Any track bounces that are contemplated can be worked out in advance. Where possible, avoid bouncing adjacent tracks. This can cause feedback across the sync head of the tape machine.

It is good practice to set the mixer flat before recording. Turn all the faders down. Turn all the aux sends off. Set the EQ flat and the routing buttons up. This will ensure that you are not inadvertently sending signals to outputs and will avoid confusion.

INPUT

Microphones are plugged into the mixer via the XLR sockets. This input will also accept keyboard instruments. If you wish to plug guitar or bass guitar direct into the board, you will need a DI box. This matches the impedance of the instrument to the mixer. Studio quality condensor microphones can be plugged in without the need for an external power supply. In this application you will need to switch on the phantom power both at the power supply and on the relevant input channels. The line input will accept the output from a tape machine, a mixer or some of the higher output electronic keyboards.

The mic/line button on the input channel determines which socket is connected.

SETTING THE INPUT LEVEL

Different microphones will produce different output levels. Similarly, the same microphone will produce radically different levels when used on a guitar stack rather than a flute.

The GAIN control is used to set the optimum level for the input channel. In order to measure level we need to send the input to a meter. The easiest way to do this is to use the PFL button above the input fader.

PFL stands for Pre Fade Listen. When a PFL button is pressed, the signal from that input is sent to the monitor channel bargraph. Because the PFL is derived before the input fader, PFL levels are un-affected by the fader settings.

When the PFL is activated, it also routes the signal to the STEREO MONITOR and HEADPHONE outputs. The control room monitor level is determined by the STEREO MONITOR gain on the monitor channel. The DIM switch above it provides an instant cut in the monitor level of 20 dB. Immediately above that there is a headphone volume marked PHONES.

When no PFL's are in use, control room monitor is derived from the main stereo output. The MONITOR BARGRAPH will then show the summed stereo monitor output.

The bargraph itself can be switched to PPI (peak program) or VU (volume unit). For the time being, it will suffice to leave the meters switched to VU. This will produce a similar characteristic to the meters on the multitrack machine. The significance of PPI versus VU will be discussed in the more advanced chapter.

Now all we need to set the gain is a representative input level. Encourage the musicians to play the actual part that will be committed to tape. If he is just 'doodling', it is probable that the input level will not be representative.

Set the gain control so that the meter shows a peak level of 0dB. If it is not possible to achieve this within the first three quarters of the gain pots travel, use the PAD. The PAD switches in a level of cut 20 dB.

As a secondary check, there is a PEAK LED indicator on each channel. This is set to flash at + 20dBm, which is 2dB before the onset of distortion. If the LED is flashing more than sporadically, the gain should be reduced.

Should you be unfamiliar with the procedure detailed, it is worth trying a little experimentation. Deliberately overload the mixer by turning the gain up too far. You will hear clipping as the signal level exceeds the headroom of the mixer.

Now try setting the gain far too low. You will find that the residual hiss of the mixer becomes too prominent.

The name of the game is to get as much level as possible without distortion.

EQUALIZATION

Controls:

By tradition the tone controls on a mixer are called equalization or EQ. Originally, equalization was created to compensate for frequency losses caused by long cable runs. As engineers discovered creative uses for EQ, tone circuits started to become more comprehensive.

The EQ on the Dynamix splits the audio spectrum into four ranges. Each of these areas can be boosted or cut by up to 15dB.

The accompanying diagrams show the equaliser curves. The vertical axis shows the cut or boost of 15dB. Frequency is shown along the horizontal axis.

The BASS control is a bell response which is centred at 60Hz. This gradually diminishes in effectiveness, up to 1kHz.

A bell response is also used for the MID band controls. Both MID bands incorporate sweep frequency controls.

The LOW-MID sweeps from 100Hz up to 1K4KHz, overlapping this is the HIGH-MID, which travels from 600Hz up to 10KHz.

The TREBLE control utilises a shelving response which diminishes in effect below 12KHz.

In combination, these four bands give a comprehensive coverage of the audio spectrum.

Below the EQ section there is an EQ BYPASS. This cuts the EQ section out of the circuit completely and allows instant comparison with the original signal.

USAGE

EQ is an invaluable tool for 'find tuning' the sound of a voice or instrument. Whether you use it when recording onto multitrack or prefer to use EQ only on mixdown, is a matter of personal choice.

Excessive equalization is to be avoided. Not only will it create a 'muddy' sound, it will also cause phase anomalies in the mix.

When there is an obvious deficiency in the original sound, go back to the source.

Consider the basic sound of the instrument. Try altering the position of the microphone or changing to a different type of microphone. Once the sound is fundamentally good, you can use the EQ to add a little 'garlic'.

Try to consider each sound as part of the overall mix. Use the EQ to give each sound its own identity. Consider, for instance, if the bass guitar can be differentiated from the bass drum.

EQ can also be used in a different way; for removing unwanted noise. The BASS control is centred at 60Hz. It is no co-incidence that this is the the frequency of mains hum. Similarly, hiss from synthesisers or amplifiers can often be reduced by selective cutting of the high frequency.

If you have adjusted the EQ, the chances are that you have altered the level. Re-check against the PEAK LED and the BARGRAPH.

Now, you have two basic options:

1) You only wish to use one channel at this point. Proceed directly to the section on routing.

2) You wish to use more than one input channel. Switch out the PFL on the channel you have been using and proceed to the next one, using routines above.

ROUTING

So far we have been monitoring the signal via the PFL circuit. The next thing is to route the input to one of the output channels.

The routing system is very simple and works in conjunction with the pan control.

Any input can be sent to any group or left and right output. Pressing more than one button will send the input to more than one output.

The PAN control sweeps between the LEFT and any odd numbered output on one side and the RIGHT and any even numbered output on the other.

In the 12 o'clock position, the pan control sends to LEFT and RIGHT, odd and even equally.

As an example, imagine that the 1 - 2 ROUTING button has been depressed, pan hard left and the input will be routed to group output 1. Panning hard right will send the input to group output 2.

A brief word of CAUTION.

We have already established that the PAN control sweeps between odd and even numbered outputs. This becomes relevant if you are using a number of inputs to record a stereo track onto two tracks of the multitrack recorder.

Imagine that you have several microphones on a drum kit. If you make a stereo mix onto, for instance, tracks 7 & 8, everything is relatively straightforward. The inputs for the drum kit would have the 7 - 8 ROUTING buttons depressed. A stereo image would be created by panning each mono input to a different place between the 7 - 8 outputs. This is exactly the way a stereo image will be created on the final mixdown.

Suppose however, that you had decided to record the drum kit onto tracks 6 & 7. This means that it would be impossible to use tracks 5 or 8 at the same time. Anything you tried to record on tracks 5 or 8 would also have the drum mix on it. The second problem would become fully apparent on mix down.

Because track 6 is on the right of the PAN control and 7 on the left, the stereo image would be reversed. On mixdown the right hand frim track would have to be panned left. The left hand drum track would have to be panned right. Whilst this is not an insurmountable problem, it is confusing. The simple lesson is this:

When using subgroups in stereo, think Left is odd, Right even.

If you do find yourself in a difficult situation you can always switch the output leads round at the back of the mixer. Don't forget to put them back afterwards.

Having routed the input to the subgroup, the input fader and the subgroup fader will determine output level.

To hear what is happening at the subgroup, switch out all input PFL's, and use the subgroup PFL. The level of the subgroup PFL is dependant on the level from the input fader.

Once again, we come back to our two basic possibilities.

If you are sending one input only to a subgroup - all you need to do is get the optimum level onto the tape machine. Faders should be at approximately three quarters of their throw and the input gain set so that the bargraph peaks at 0dB.

You may be sending several inputs to one or more subgroups - as in a stereo recording of a drum kit. In this instance you will want to create a musical mix between the inputs. The input faders will be used for this purpose. Level onto tape will be controlled by the subgroup faders.

If you have been following this chapter step-by-step, you probably have a lot of musicians waving through the window. As yet we have not provided them with any monitor.

FOLDBACK MONITOR - USING THE AUX 1 & 2 CIRCUITS

Aux 1 is pre-fade auxiliary send. In other words, it provides a separate mix that is un-affected by the fader settings. Aux 2 is identical and provides a second mix. Not all musicians will want to hear the same monitor mix. A channel is only connected to Aux 2 when the Aux 2/Aux 3 button is up. When the button is down, that control is routed to the Aux 3 circuit.

Providing that Aux 1 and Aux 2 outputs are connected to headphone amps, the rest is easy.

The Aux 1 and 2 controls across the input section will provide a mix of the input channels. Master level to the foldback amp is on the MONITOR MODULE.

To hear the mix you are creating through the control room monitors, press the Aux 1 or 2 AFL (After Fade Listen) button.

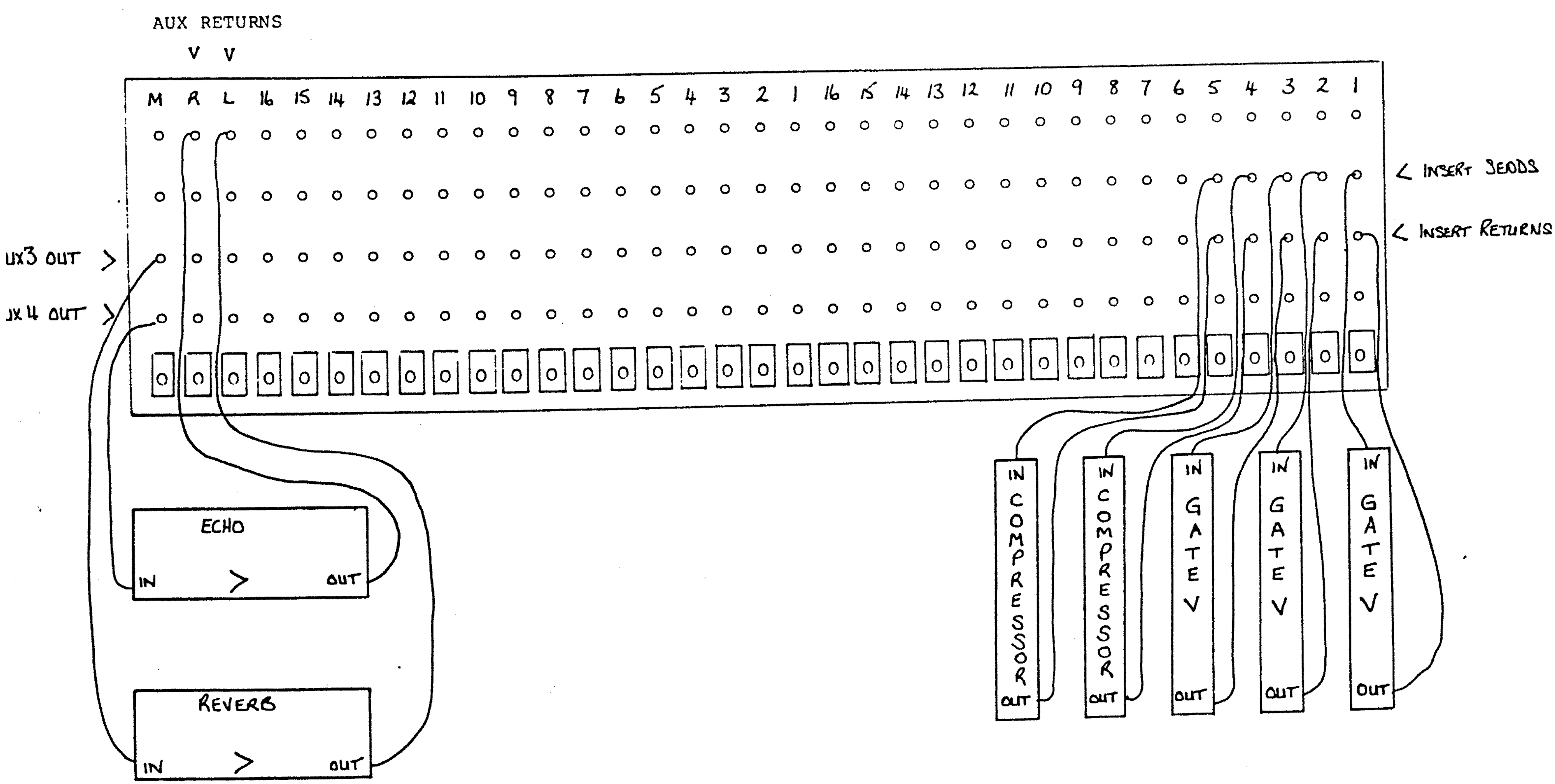
USING THE TALKBACK FACILITY

Plug a mic into the TALKBACK socket and you can communicate with the studio floor. There is no particular impedance requirements for the mix. An omni-directional is probably more useful as it saves stooping over the mic.

The talkback level is set by the TALKBACK volume control and the output is fully routable. When any of the talkback routing buttons are used the stereo monitor output is muted to prevent feedback.

To get a two way conversation going, press the talkback routing for Aux 1 and 2. This will route the talkback mic to foldback. At the same time PFL one of the mics in the studio.

Some effects patches with the D3000



By routing the talkback to SUBS, you will be able to 'mark' each take on the tape.

EFFECTS - USING THE AUX 3 & 4 CIRCUITS

Where possible, some producers prefer to add effects on mixdown only. However, there is a school of thought that believes it is important to commit the intended sound to tape immediately.

Occasionally effects have to be recorded onto multitrack. For example, a unit such as a harmoniser may be required to perform two separate functions simultaneously during a mixdown. One of these effects will therefore need to be committed to tape first.

Additionally, it will be necessary to record effects onto the multitrack if you are bouncing tracks. Once a number of separate tracks have been combined it will not be possible to single one of them out for effects treatment.

Aux's 3 & 4 are post-fade sends. This means that reducing the fader level on a given channel will also reduce the level going to the Aux 3 and 4 circuits. As a result, the ratio of 'dry' signal to effect will remain constant for a given channel, regardless of changes in fader level.

The master output levels for the Aux 3 & 4 circuits are on the MONITOR MODULE. An AFL (after-fade-listen) button is provided. Depressing one of these will allow you to hear the Aux 3 or Aux 4 in isolation.

Aux 3 and Aux 4 OUT sockets on the back of the monitor module should be connected to the inputs of the effects that you wish to use.

Effects may then be returned via the AUX RETURN sockets on the OUTPUT MODULES.

These have a PAN and ROUTING system, identical to the one found on input modules. It is therefore possible to assign an effect to a subgroup, for recording onto multitrack, or to the left and right outputs for mixdown.

Care should be taken not to overload effects units as this will cause distortion. Equally, inadequate levels will cause a deterioration in the signal-to-noise ratio.

The PFL monitor on the AUX returns will allow you to check the signal from an effects unit. In addition, most effects units have a level indicator or overload LED fitted to them.

Effects with a stereo output may be returned via both AUX RETURN sockets. Of course, if you are recording onto a single track, the end result will still be mono. Nonetheless, it may still be worth returning both effect outputs. With some chorus devices you will leterally lose half the effect if you don't.

Any LINE IN on the mixer may be used for effects return. Using a spare input channel has the added advantage of providing EQ for the effect return.

If you do decide to use an input channel in this way, remember not to turn up the Aux 3 or Aux 4 level on that channel. This will only create a feedback loop.

INSERT SEND AND RETURN

In order to extend the flixibility of the mixer, 3000 series consoles have and INSERT SEND and INSERT RETURN socket on every input and output channel. This allows for the insertion of a line level device. INSERT SEND and INSERT RETURN level controls are provided so that level adjustments need not be made from the outboard equipment.

These inserts are extremely useful when an effect is to be applied to one instrument only.

EFFECTS ON FOLDBACK

Effects may be introduced to the Aux 1 and Aux 2 circuits. It often helps the musicians to have a little revern or echo on monitor. This takes away the dead sound of the recording room. Vocalists and lead players will find this particularly helpful.

This is the easiest way to use effects on foldback WITHOUT comitting the effect to tape:

1) Connect the effect to the output of Aux 3 or Aux 4 as usual.
2) Return the output of the effect via the LINE INPUT of an un-used input channel.
3) The LINE/MIC button on the input should be set to LINE.
4) Using the input channel PFL, set the input gain to the optimum level. (As per usual, you will need the musicians to provide a realistic level). You will also need to set the Aux 3 or Aux 4 send levels at the same time.
5) Changing now to the Aux 1 or 2 PFL, use the Aux 1 or 2 gains on the input channel to set the overall effects level.

6) Do not turn up the Aux 3 or 4 level on the input channel as this will cause a feedback loop. Of course, if you are using Aux 2 for monitor, you will be obliged to use Aux 4 for the effect.

7) Avoid routing the input channel or using the input fader. You may inadvertantly introduce the effect to tape.

OVERDUBS

The procedure for overdubbed tracks is fundamentally the same as for recording the first track.

This time however, the musicians will need to hear both their own instruments and the pre-recorded tracks.

Live input goes through the input channel Aux 1 or 2 as normal. The pre-recorded tracks will be introduced to the Auxes via the sub-groups.

The TAPE MON button on each subgroup switches in the off tape monitor for the reciprocal track on the tape machine. This is introduced to the foldback via the GROUP TO AUX 1 and GROUP TO AUX 2 level controls.

As a practical example, let us imagine that tracks 1, 2 and 3 have been recorded already.

First we depress the TAPE MON button on subgroups 1, 2 and 3. Secondly we introduce the pre-recorded tracks into aux mixes via the GROUP to AUX level controls.

To check the AUX mixes we need only depress the Aux 1 or Aux 2 AFL buttons on the monitor section.

It is worth setting up as good a monitor mix as possible. If the monitor mix sounds good and punchy, it will inspire the musicians to try harder.

It is possible to create a monitor mix using only the subgroups. Input channels that are routed to subgroups can be monitored via the subgroup monitor section. The TAPE MON buttons will have to be up on these channels.

This technique does have a disadvantage if more than one input is sent to a subgroup. The mix at the subgroup will be the same as the monitor mix for those inputs.

Note that the subgroup TAPE MON circut has no effect on the input channel AUX sends. This means the same information may be introduced to the foldback mix from two places, i.e., when TAPE MON buttons are set to monitor group outputs, GROÚP TO AUX controls will feed information that is already available at the input channel.

Nothing particularly terrible will happen if you have aux feeds from two places. It's just that some inputs will have two volume controls.

DROP-INS

There are often times when 90% of a track is very good. The remaining 10% however, renders the track unusable. Rather than re-record the whole track it is possible to 'drop-in' a small section to repair the damage.

Vocal tracks often utilise extensive 'drop-ins', to help capture the optimum performance.

There are a number of factors that will help to ensure a seamless 'drop-in':

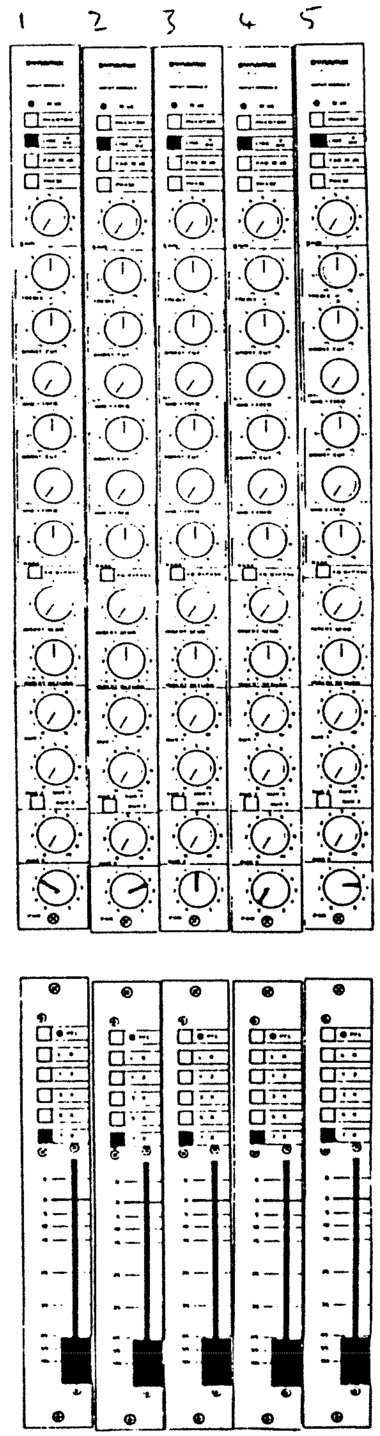
Firstly, it is advisable to pick a 'drop-in' point that represents a natural break in the music. The end of a verse or a vocal line can be considered. It is equally important to pick a convenient point to 'drop-out'. If you overshoot the selected 'drop-out' point, you will start to record over the O.K. section.

The second consideration involves input settings. If the input level, EQ or effects settings are different, you will certainly hear the join.

When dropping into a track, you will need the TAPE MON button down on that subgroup monitor. In addition to monitoring off tape, you will also need to monitor in sync. (Not all multitrack tape machines have separate sync and replay heads. If yours does, you'll need to be in the sync mode).

BOUNCING

Track bouncing is the standard technique for increasing the capacity of a multitrack format. Transferring tracks in this manner should be thought of in the same way as a full mix down. Decisions made regarding the mix of the bounced tracks cannot be reversed once the original tracks have been recorded over. Similarly, if effects are to be added

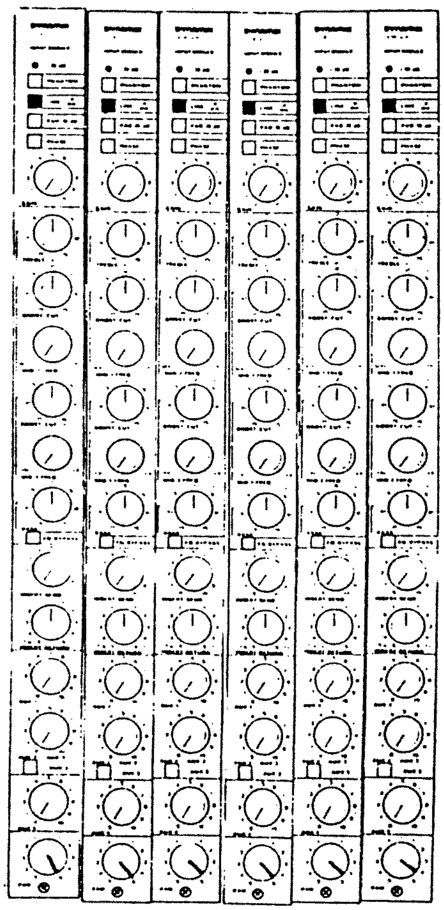


Bouncing tracks 1 - 5 onto tracks 7 and 8.

□ = buttons in up position

■ = buttons in down position

Pan controls are adjusted to taste, so that tracks 7 and 8 form a stereo recording on replay



Bouncing tracks 1 to 6 onto track 8

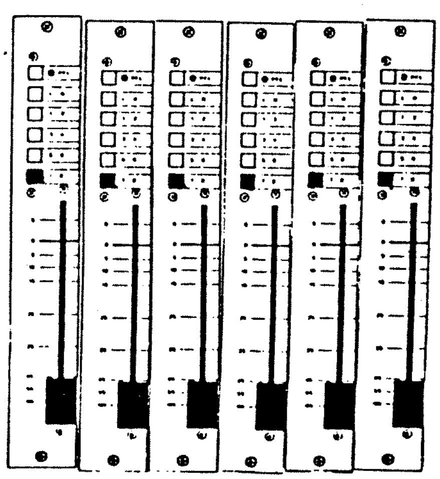
□ = buttons in up position

■ = buttons in down position

Individual EQ should be set up before bouncing tracks (see notes below on effects).

Think about any effects you may wish to add, as it will not be possible to single out individual instruments after their tracks have been bounced together

All pan controls should be hard right. (To bounce onto track seven they would need to be hard left).



The balance between the tracks is created by the input faders. Level onto tape is set with the group faders.

to individual tracks, this must also be done at the bouncing stage.

On the adjoining page, there are two diagrams, showing typical bouncing configurations. In the first diagram, tracks 1 to 5 are being bounced into stereo onto tracks 7 and 8. The second example depicts a mono bounce of tracks 1 to 6 onto track 8.

Tracks which are to be bounced must be put into remix by switching the corresponding input channels to LINE.

It is advisable to re-set the input gains by PFL'ing each channel to the MONITOR BARGRAPH.

Each input must now be routed to the relevant subgroup/s, using the PAN and ROUTING controls. Level and mix to the subgroups can be checked via the GROUP TO OP GAINS. These introduce the subgroups to the left and right outputs and also the control room monitor. Alternatively, use the group PFL's.

The mix can bow be set up, using the input faders. Level to tape is controlled by the subgroup faders.

If you intend using effects on the bounce, remember to route the AUX returns to the relevant subgroups, not the L-R outputs.

As mentioned in the section on TRACK PLANNING, it is best to avoid bouncing adjacent tracks. This can lead to feedback across the sync head and places limitations on the levels that can be used.

If all the information on tape is to be bounced, there is no reason to bounce in sync. Bouncing via the REPRO head will eliminate the feedback problem and may improve the frequency response.

Alternatively, it is possible to fill all the tracks on the multitrack recorder.

These can then be mixed down onto the stereo machine. The stereo mix is then dubbed onto two tracks of the multitrack machine. As this involves two bounces, some people prefer not to use this technique.

MIXDOWN

The quality of the final products is dependant on the mixdown to a large degree. It is a job that should not be undertaken in a hurry. Similarly, where possible, it is best to avoid starting a mixdown at the end of a long recording session. After a time the ears become tired and it is difficult to make good artistic judgements.

If you are tunning a commercial studio, it is a good idea to explain the importance of a mixdown when booking artists. A really good mix can take anywhere between a third and half the total recording time.

Too often bands cram a mixdown into the last hours of a session. What could have been a top quality recording ends up as 'just another demo tape'.

ROUTING AND CONNECTIONS

When re-mixing the inputs corresponding to the multitrack machine will be switched to LINE.

Inputs may be sent direct to the left and right outputs. Alternatively, sections of the mix can be routed to one or more subgroups. This will provide an overall level control for, say the drum tracks.

Subgroups are routed to the main left and right outputs via the GROUP TO OP levels and the PAN controls.

As there is no longer any need for foldback, all four Aux busses may be used for effects send. Remember though, Aux 1 and Aux 2 are pre-fade. The effects levels will not be controlled by the input faders.

Additional effects sends can be taken from the subgroup outputs. These may be returned via unused input channels.

Outboard devices that are to be used on one track only can be connected via the INSERT SEND and INSERT RETURN sockets on the input channels. Level for these devices is controlled by the INSERT SEND and INSERT RETURN gains on each channel.

Groups of tracks may be treated in the same way, using the INSERT SEND AND RETURN facilities on the subgroups.

The same facility is found on the main outputs and this is particularly useful for a device such as a stereo compressor limiter.

PAN

Most of the tracks on the multitrack recorder will be mono. The only exceptions will be instruments that were recorded in stereo onto two tracks, such as a piano. You may also have two tracks that contain a stereo bounce. It will be necessary to pan a stereo pair of tracks left and right to retain a stereo image. The further apart you pan the two tracks, the wider the stereo image will be.

The remaining mono tracks will be panned to varying positions in the stereo mix. Exactly how you pan the tracks depends on the effect that you wish to create.

A drum kit, for instance, will sound more 'real' if the individual drums are panned to their normal positions, i.e., snare and hi-hat to the right, bass drum in the middle etc. However, panning the tom-toms further apart than usual will create a greater sense of movement when the drummer goes round the kit.

Instruments may be panned during the mix. A solo could drift from the left side to right. A succession of staccato chords could be sent left and then right in rapid alternation.

All this is a matter of artistic judgement; there are no right or wrong solutions.

E.Q.

Equalization is not just a question of making things sound 'better'. Part of the exercise involves making it easier for the listener to differentiate between the various instruments.

For this reason it is advisable to always consider equalization changes within the context of the total mix. You may find it easier to set up the equalization by listening to one channel at a time. Having done so, listen to the effect it has on the overall sound.

The EQ BYPASS will enable you to make instant comparisons with the unaltered track.

EFFECTS

There is great potential for using effects to enhance the sound on mixdown. Many of the effects can be used to make the sound 'bigger'.

Effects, such as stereo reverb plates, should be returned via both AUX RETURNS or two input channels. The widest stereo panorama will be obtained by panning hard left and right.

Mono effects can still be used creatively in stereo. Mono tracks can be 'fattened up' by using a mono delay and panning the return the other side of the stereo image.

KEEP IT CLEAN

Often, there will be intervals of un-recorded tape on a particular track. This is especially true at the start of a song.

Rather than adding tracks containing nothing but tape hiss, it makes more sense to cut them out of the mix altogether. Providing you are familiar with the arrangement, this is fairly easy to achieve manually.

When routing buttons are all up, the input is not being sent anywhere. After a few trial runs you should be able to switch tracks in only when they are needed.

MIXDOWN LEVELS

Level to the stereo tape machine is controlled by the master output faders.

Playback from the mixdown machine is facilitated by the STEREO TAPE TO MONITOR button on the monitor section. Playback level is controlled by the STEREO MONITOR gain.

CONCLUSION

If you have followed this chapter through, you should now be able to complete a multitrack session.

In most instances, only one way of achieving a result has been explained. As the D3000 is an extremely comprehensive mixer there are often several ways of achieving the same result.

The next chapter discusses some more advanced techniques, to help you explore the creative possibilities of the Dynamix mixer.

MORE ADVANCED TECHNIQUES

Contents:

PPI VERSUS VU

The Dynamix 3000 is switchable between two types of meter response. These are PPI (peak program indicator) and VU (volume unit).

In the VU mode, the ballistic response of the bargraph will be similar to the VU meters on a tape machine. Put simply, they will move up and down at about the same speed.

As VU's are relatively slow moving they are easy to read and a useful indicator that the output from the desk is consistant with the operating levels of the tape machine. For this reason, the VU mode is calibrated so that 0 VU = +4dB RMS, which is the standard level for recording studio equipment.

Between the nominal operating level of the equipment and the onset of distortion there is a certain amount of leeway, known as headroom.

Because VU response is not very fast, it is not a very good way of determining how much of the available headroom is being used. A fast transient peak, such as a snare drum produces, is simply too fast for the VU to follow.

PPI response is extremely fast. It will follow fast transient peaks and is a reliable indicator of the amount of headroom being used.

There is a secondary scale inscribed on the bargraph for the PPI mode and this goes up to +14dBm to accommodate the peaks that cannot be tracked on a VU response meter.

For broadcast applications, PPI metering is used exclusively. It is essential to monitor headroom as overloading the transmitter violates broadcast regulations.

In the recording studio, it is often used as a secondary check on more percussive signals.

NOTE:

> Japanese multitrack equipment often uses a -10dB operating level. The Dynamix 3000 can be converted to -10dB standard. Please consult your dealer for further details.

BRIDGING EQ UNITS TOGETHER

As the equalization section of the Dynamix 3000 is very comprehensive, it should suffice for 99% of uses. However, on the odd occasion when you need more EQ there is a technique which does not involve outboard equipment.

The INSERT SEND socket on an input channel of the 3000 is wired post EQ.

By plugging the INSERT SEND of one channel into the LINE INPUT of another channel, you will link up two EQ circuits. This creates an EQ section with four mid-range sweeps and a possible cut and boost of 30dB! Be warned, this is quite enough to destroy some speaker drivers.

MORE FOLDBACK CIRCUITS

Just as Aux 1 and 2 will double as effects sends, Aux 3 and 4 can be used as additional monitor circuits. Of course, changing fader settings will alter the Aux 3 and 4 mix, but this is not too much of a problem if you set the faders before you set up the foldback mix.

Any line level output can be used for foldback providing:

a) the resultant mix is satisfactory to the artist
b) the output is not presently employed fulfilling another function

MORE USES FOR THE INSERT POINTS

Insert points will accept any line level device and may be used with a wide range of effects. Their most useful applications are often with compressors and noise gates.

Compressors are widely used, both as a device for recording the maximum level onto tape and also as an effect. During recording they are seldom used on more than one input at a time. The insert sockets are therefore a more appropriate patching point than the auxiliary circuits.

Stereo compressors are also used to limit the dynamic range of a complete mix. This is often done to make the track more 'commercial' for airplay purposes. The limited dynamic range of radio stations and the popularity of radio as a supplement to existing noise has tended to increase the desirability of the constant output levels.

The INSERT SEND and RETURN sockets of the OUTPUT MODULES will accommodate a stereo compressor without the need to disconnect the master recorder.

Noise gates are used in two main contexts. Firstly, they are used during recording, either to switch out noisy instrument amps or to minimise spillage on the studio floor. Secondly, they are popular on mixdown. This prevents tracks that have little on them, such as a few bars of solo, adding to the noise level the rest of the time.

Once again, the INSERT SEND and RETURN sockets alleviate the need to disconnect other equipment.

USING EFFECTS WITH STEREO INPUTS

In the first recording chapter, it was explained that effects with stereo outputs could be disconnected to both AUX RETURNS.

These could then be panned left and right to enhance the stereo image.

To get the maximum effect from a device that is stereo input and output it would be necessary to creat a stereo send.

It would be possible to use the main outputs for this purpose. However, as all inputs would receive the same level of effect, this would not normally be very usable. Bass instruments sound very muddy when put through echo or reverb. There is a good chance that the drum kit would cause overload.

An alternative would be to take the output from two of the subgroups. In some applications this will be acceptable. This method would allow selected inputs to be sent to the effect unit. Unfortunately, it would not be possible to put, as an instance, more reverb on the backing vocals than on the main vocal.

The most acceptable option would be to use both the Aux 3 and Aux 4 mix. By using one auxiliary for the left channel and the other for the right it will be possible to create a full stereo send. (after all, a pan control is only a device which controls the left and right volume in inverse proportion to each other).

JOINING THE BEST TWO TAKES TOGETHER

A situation often arises in a session where two takes of the same part have been recorded on separate tracks. It then becomes necessary to decide which is the better take. This is not always an easy decision. Sometimes parts of one take are better than the other and vice versa. Sooner or later someone asks 'Couldn't we just join them together?' Providing that there is a track spare the answer is usually 'Yes'.

The success of this type of 'edited bounce' depends on a number of factors. Firstly, the volume and other settings on the two tracks must be as similar as possible. Secondly, the engineer must know exactly which bits are to come from which track. Lastly, the engineer.needs a steady hand and a good sense of rhythm.

The technique is much the same as for a regular track bounce. However, the level to tape needs to be set on the basis that only one track will be playing at a time. Monitoring is best done from the subgroup as it is difficult to tell if the edits were successful with both tracks playing at once.

Unlike 'drop-ins', this technique can be repeated as many times as it takes to get it right. Simply switch the routing buttons in and out to send the desired track to tape. The art in all this is not so much remembering where to change track; its changing just as the last note dies and the next one is about to come in.

ADVANCED TRACK PLANNING

The problem with allowing a track to evolve organically is simply this: if you start to run out of tracks, it is normally the more essential ingredients that get bounced first. In an extreme example, the rhythm tracks will be three generations old whilst the backing vocals are direct onto tape.

This can be avoided if the arrangement is already worked out before recording starts.

First a guide track is layed onto one track of the tape. The precise content of this is not vital providing:

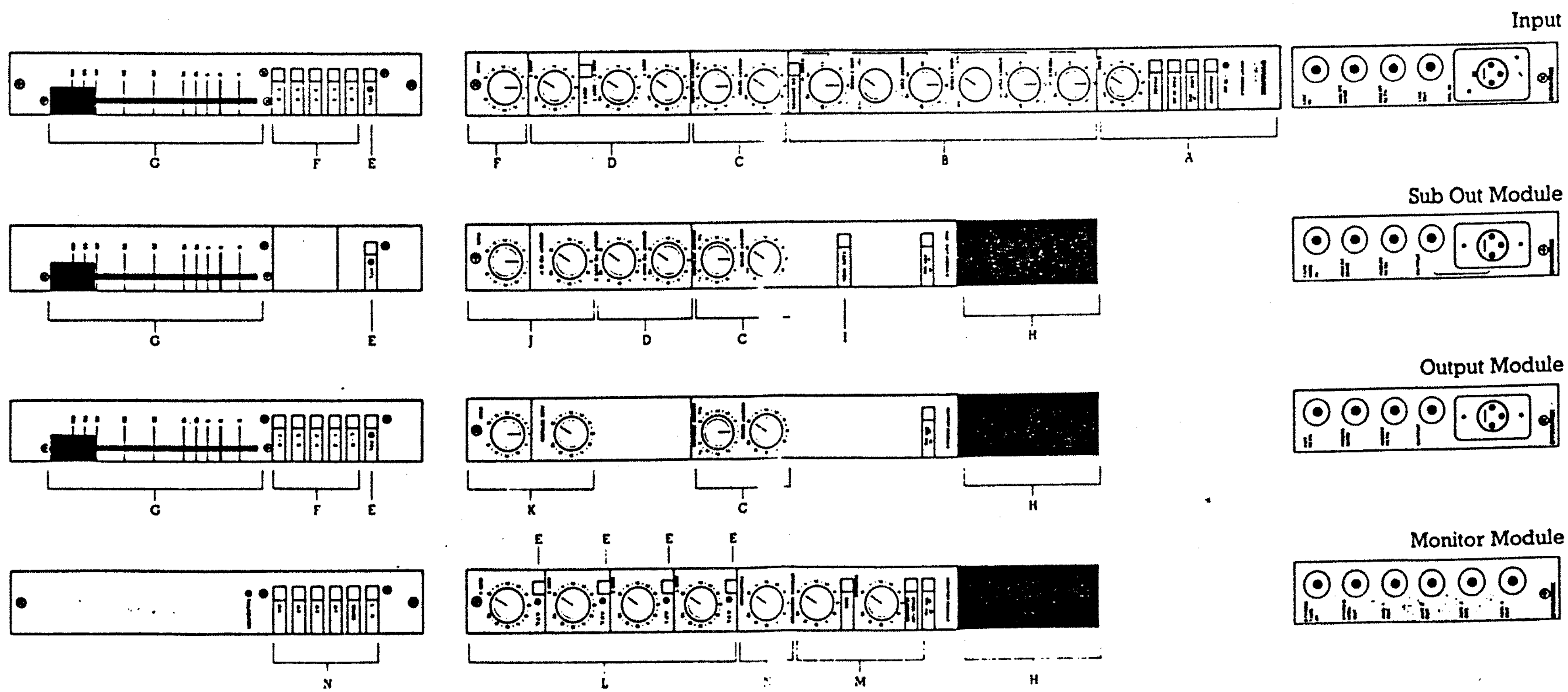
a) it is rhythmically exact
b) it is in tune
c) it is possible to tell exactly where you are on the track

Drum machine, guide vocal and piano would be a workable combination.

Now the parts are recorded in reverse order of priority; the multitracked 'ooh aah' vocals, the triple tracked string machine, the six rhythm guitars for that really 'fat' sound. These can all go on first and be bounced down to two or three tracks.

You should be left with sufficient tracks to get the main instruments and vocals down without further bouncing.

Mic/Line Input Section A
Equaliser Section B
Insert Section C
Auxiliary Section D
PFL/AFL E
Routing Section F
Channel Fader G
Bargraph Displays H
Group/Tape Selector I
Group to Output Section J
Auxiliary Returns K
Auxiliary Send Master Controls L
Headphones and Monitor System M
Talkback System N



STARSOUND DYNAMIX

MIC/LIN INPUT SECTION A

The MIC INPUT is via an XLR type connector wire Pins 1, screen 2, return 3, live and is electronically balance. The input may be used with all types of dynamic microphone of low impedance output, or with condenser microphones using the internal 48V PHANTOM POWER supply. The LINE INPUT via stereo ¼" jack is also electronically balanced, but will accept a mono plug carrying an unbalanced signal. This input is high impedance and will match to almost any line level input.

The MIC-LINE SWITCH selects either of these two inputs, or acts as a 20dB attenuator on the mic input if nothing is plugged into the line input socket.

The PAD SWITCH when depressed attenuates both mic and line inputs by 20dB enabling the front end of the mixer to be matched to the widest possible range of input signal levels to avoid input overload.

The PHASE SWITCH can be used to reverse the phase of both the mic and line inputs. This is particularly useful with multiple microphones on a sound source to correct for 'dead spots' due to phase concellation between two or more microphones.

The GAIN CONTROL on a 41 position detented potentiometer varies the gain of the input amplifier over a range of 30dB and should be used in conjunction with the pad so that it is between 40% - 80% of maximum. The input gain should be set as high as possible without causing the peak indicator to illuminate, so as to ensure the best possible distortion and signal to noise performance of the mixer, and should not be used as a channel volume control.

The PEAK LED situated above the input selector switch bank flashes when the pre-fade signal level peaks at +20dBm (i.e. comes within 2dB of clipping). This signifies that the input gain should be reduced slightly.

EQUALISER SECTION B

This section of the mixer consists of fixed frequency Bass and Treble and two parametric mid range filters. All of the boost and cut potentiometers are centre detented for easy zeroing, and the two parametric sweep potentiometers are continuously variable. The equaliser section may be bypassed completely by depressing the EQ OFF SWITCH, to allow A - B comparison of the treated and untreated signal.

The two overlapping parametric mid range filters allow an enormous range of tonal control over the signal enabling the sound engineer to achieve clear separation of signals in the mix, as well as leaving the bass and treble controls free to compensate for system deficiencies due to microphones, loudspeakers, room acoustics etc.

EQUALISER CHARACTERISTICS

Treble	+/- 15dB	@	12KHz	- shelving response
Mid 1	+/- 15dB	@	10KHz	- 600Hz sweepable - bell response
Mid 2	+/- 15dB	@	1.4KHz	- 100Hz sweepable - bell response
Bass	1/- 15dB	@	60Hz	- Bell response

INSERT SECTION C

Primarily the channel insert is intended as a facility to patch effects into an input or output channel, but by adding a SEND LEVEL control and a RETURN MIX control this input output facility is given a much wider range of uses, as well as extending the versatility of its effects usage.

When used as an effects insert, the output to the effect unit is taken from the mono ¼" rear panel INSERT SEND jack and the level is adjusted by the 41 position detented insert level control, allowing a wider than normal range of ancillary units to be used. The output from the effect is connected back to the mixer via the rear panel INSERT RETURN jack. The centre detented insert return mix control may be used to balance the proportion untreated signal (sig) to the desired proportion of treated signal (Fx) from 0 to 100% either way.

The insert send output may also be used as an independent output to drive for example a multitrack tape machine for live recording, or a stage or studio monitor direct or via a second mixer.

The insert return may be used as a line level input if more channels are needed than available.

AUXILIARY SECTION D

Four auxiliary sends are available on input channels.

Auxiliary 1 and 2 are pre fade for use as foldback channels and Auxiliary 3 and 4 are post fade for use as effects sends. On input channels one level control is switchable between Auxiliary 2 and pre fade send, and Auxiliary 3 and post fade send to allow the user to decide whether he requires two pre fade, or two post fade auxiliaries, without limiting the use of the two pre fade auxiliary sends from the subgroups for use either as foldback or stereo studio floor monitor outputs.

PFL/AFL E

When a PFL or AFL button is pressed, the signal related to that switch is 'soloed' on the stereo monitor and headphone outputs and the signal level is displayed on the monitor channel bargraph. There is a red LED by the side of every PFL and AFL button that illuminates if the switch is depressed.

All other mixer outputs are unaffected by the PFL system. The PFL system is particularly useful for 'setting up' input channels and also for tracing system faults.

ROUTING SECTION F

An input channel or auxiliary return may be routed to any pair of sub-group outputs, or direct to the master left and right outputs while the pan control pans between the left (odd) and right (even) busses.

In a PA application this allows the subgroups to be used as stereo submasters for groups of inputs, while in the studio, channels can be assigned to single subgroups and multitrack channels while still allowing direct access from inputs to the master outputs for mixdown.

CHANNEL FADER G

The channel fader is a high quality 100mm travel Alps fader, scaled from to 6dB. It controls the output level from the channel to the selected busses on inputs, and to channel outputs on input, subgroup and main output channels.

BARGRAPH DISPLAYS H

The 25 segment LED bargraph display indicates the output signal levels on subgroups, main outputs and the stereo monitor output in two switch selectable modes.

The VU mode has a VU type ballistic response and is calibrated to 0VU=+4dBm RMS (recording studio standard) while in the PPI mode it has a peak programme indicator response calibrated to +4VU=+20dBm this mode is to ensure that fast transient peaks that cannot be tracked on a VU response meter do not go into overload.

GROUP/TAPE SELECTOR I

The group/tape switch selects the subgroup input between the subgroup mix amp output and the tape return input. The primary use of this facility is for tape monitoring purposes during multitrack recording. The tape tracks allow the full input capability of the mixer to be available for recording subsequent tracks. When a subgroup is being used as a tape monitor input, that subgroups mix amp output may still be accessed on the insert send output, for use as an effects send or independent monitor output.

Other uses for the group/tape selector switch are as a subgroup mute, or as a facility to 'drop in' effects into the subgroup signal path by taking an output from the insert send and returning the signal from the effects unit into the tape return input.

GROUP TO OUTPUT SECTION J

This consists of a 41 position detented level control and a centre detented pan control which feeds the left right output busses with the post fade subgroup output signal. In a studio application this section would be considered as the stereo control room monitor send, while in a PA system this is the subgroup to the main outputs level and pan.

AUXILIARY RETURNS K

Each master output has an auxiliary return input channel intended for use as an effects return input from Auxiliary 2 and 3 outputs, but they may also be used as additional line level inputs. The Auxiliary return inputs have an input level control, AFL, pan and full routing for maximum flexibility.

AUXILIARY SEND MASTER CONTROLS L

Each of the four auxiliary send channels has a master output volume control and associated AFL button for output signal monitoring. Auxiliary send outputs are on ¼" rear panel jack sockets.

HEADPHONES AND MONITOR SYSTEM M

The headphones and stereo monitor outputs have separate output volume controls on 41 position detented potentiometers.

The stereo monitor output is intended to drive the control room monitor system while the headphones output situated on the front panel will drive an output of four watts into low or high impedance phones. The signal source to the phones and monitor outputs is derived from any one of three sources, either from the PFL/AFL mix amp output or the master left right outputs, the latter being the usual, while the former is automatically selected when any PFL/AFL switch is pushed. The third source is the rear panel stereo master tape replay. Also associated with the stereo monitor output is a Dim switch that attenuates the stereo monitor output by 20dB.

The summed stereo monitor output signal is displayed on the monitor channel bargraph.

TALKBACK SYSTEM N

Talkback input is via the front panel XLR and is suitable for most high and low impedance microphones. The talkback level is set by the talkback volume control and the output is routable to any one of the four auxiliary outputs, the left right output pair or all subgroups simultaneously. When any of the talkback routing buttons are pressed the stereo monitor output is muted to prevent feedback, and an LED lights below the routing bank to indicate talkback on. A communication condition between the control room and studio may be achieved by routing the talkback to Auxiliary 1 and 2 and activating PFL on the relevant input or subgroup channels.