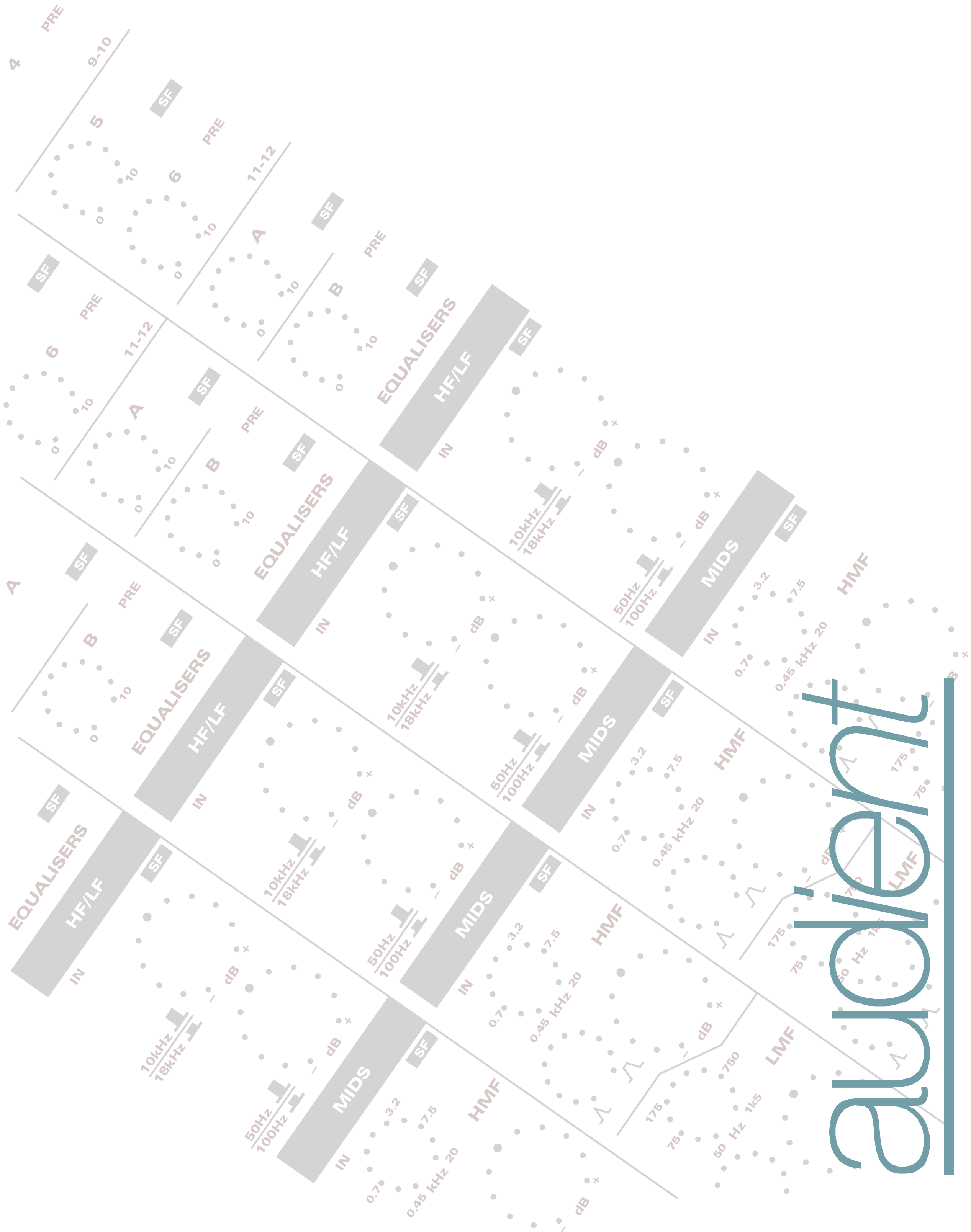
operation manual



audient

ASP 8024

Contents

Thank you for selecting an ASP8000 Series console for your application.

We have designed this equipment to provide you with the best possible tool to deal with today's demanding requirements.

We have taken a great deal of pride and care in the manufacture of this equipment so that it will provide consistent and reliable performance.

Please take a little time to study the contents of this manual so that you can be sure of getting the best performance from this equipment.

HOW TO USE THIS MANUAL

This manual has been divided into sections for your convenience.

The first section is introductory and gives a general overview of the console and its features. The following sections give a detailed explanation of the console functions, how they operate and how they relate to each other.

Please note that options such as patchbays have their own sections in this manual.

If you are new to recording then please take time to study the introductory sections which will help you understand the functions of the controls described in the later sections of the manual.

UNPACKING

Your ASP8000 Series Console has been carefully and meticulously tested and inspected before despatch.

Please check for any signs of transit damage. If any signs of mishandling are found please notify the carrier and your dealer immediately.

Your ASP8000 Series Console packing should contain an ASP8700 PSU, power cord, PSU cable and a bag of stand hardware along with this manual.

IMPORTANT SAFETY INSTRUCTIONS

Please read all of these instructions and save them for later reference before attempting to connect the ASP8700 PSU to the AC power source and the ASP8024 console. To prevent electrical shock and fire hazard follow all the warnings and instructions marked on the ASP8700.

· This unit is connected via its power cord to the mains safety earth.

NEVER OPERATE THE UNIT WITH THIS EARTH CONNECTION REMOVED

· Check that the correct operating voltage has been set for your AC mains supply

· Check that the fuse fitted is the correct type for the mains voltage selected

· Always replace fuse with the correct

THE STAND

The stand will be found beneath the console in the crate . There are 2 identical legs , a crossmember and 2 cover panels.

Lay the legs out approximately the width of the console apart and place the crossmember between them. Use the supplied hex headed M8 bolts to secure the crossmember to each leg (a 13mm AF spanner will be required). The unused holes in the outside of each leg should be filled using the 9.5mm caps supplied.

Screw a foot onto the bottom of each leg. Place the stand in an upright position and place the console on top. Secure the console with 12 M6 bolts. Finish by adjusting the feet to take up any unevenness in the floor. There is about 1 inch of adjustment available on each foot. Cover panels for the rear of the stand legs and are secured using 4 M4 screws each.

REMOVING MODULES FOR SERVICE

The console is divided into a number of removable modules. In a 36 channel console there are 3 I/O modules each containing 12 I/O Channels, a Master Module, 3 I/O Input Pods and a Master Meter Pod.

To access an I/O or Master module the wooden arm rest must be removed by undoing the 4 x M6 retaining screws from the underside of the console. Release the 2 x M6 screws from the underside of the console, located in the dimples below the module to be removed. Remove the extruded trim strips from each side of the required module and between the meterbridge and the modules.

Swing the module up (it is hinged) and remove the 3 x 40 way connectors on the right hand side of the circuit board. Remove the 3 x 16 way cables to the meter panel**.** If automation is fitted be careful to remove the ribbon cables linking each fader to the automation controller cards before attempting to fully lift the module.

Remove the green grounding cable by undoing the single M3 screw attaching it to the circuit board. Hold the module by the sides, lift it slightly back and up to clear the hinge then pull forward for removal.

Removing the Master Module is similar to the above. All ribbon cables should be removed **other than those marked LINK 1 and LINK 2**. The two power connectors should be removed noting their orientation for later reassembly.

Installation of the modules is a reverse of the above procedures. Hold the module in an approximately horizontal position making sure that it is correctly aligned over the hinges and then gently lower it into position. With the module in position reconnect the cables to their original positions.

To remove the pods the connector panel should first of all be hinged down by unscrewing the 7 x M4 securing screws. **The panel must be supported in it's lowered position to prevent it becoming detatched from the frame**. Undo the cables to the connector panels **at the connector panel end.** This gives access to the 3 x M6 screws retaining the Top Trim which can then be removed.

The 3 x M3 screws retaining each pod can now be removed. The pod will now hinge forward slightly and can then be removed by lifting it vertically up.

MECHANICAL INSTALLATION

Care should be taken not to obstruct the power supply unit's ventilation holes and adequate air flow must be provided within rack cases to prevent the unit from overheating.

NEVER PLACE THE POWER SUPPLY ON THE FLOOR AS THIS OBSTRUCTS THE AIR FLOW.

The front panel of the ASP8024 Console features a rugged under-surface printed polycarbonate overlay. Exposure to direct sunlight for extended periods should be avoided as this can have a detrimental effect on the overlay panel and on the control knobs.

MAINS POWER SUPPLY

VOLTAGES

The ASP8120 Console Power Supply features an external mains voltage selector. Switched to the 230v position the unit will operate without performance degradation from 210v to 250v. In the 115v setting it will accept from 105v to 125v. Do not attempt to operate the unit outside the ranges defined above.

For 100v operation please contact your dealer.

FUSES

Please note that the fuse ratings for the two voltage ranges are different – $1^1/_4$" T8A for the 230v setting and $1^1/_4$" T16A for the 115v setting. **Always replace fuses with the same type.** The mains fuse is very unlikely to fail under normal use and caution should be excercised if a failure should occur. Check the mains voltage setting, condition of the mains cord and integrity of the mains supply before replacing the fuse.

AUDIO INTERFACES

The ASP8024 Console has been designed and developed to provide highly robust system integration interfaces, allowing worry-free system hook-up under the most demanding situations.

Inputs and outputs are implemented using advanced electronically balanced or ground sensing topologies and are fitted with extensive RFI rejection networks. All signal interfaces are also fully protected against accidental misuse e.g. by the connection of phantom powered cables.

PIN CONVENTIONS

Signal interfaces are provided on either metal shell locking XLR, TRS Jack, 25 pin D Sub or optional 56 pin EDAC type connectors. XLR Pin 1 is connected to the chassis safety ground. XLR Pin 2 is designated as signal positive and Pin 3 as signal negative.

To unbalance the outputs of the ASP8024 Pin 3 should be connected to Pin 1 at the output of the console. Similarly, inputs from unbalanced sources should be connected via twin screened cables with the Pin 3 connection tied to the screen at the unblanced source.

SCREEN CONNECTIONS

In order to maintain optimum EMC performance it is important that screens are properly connected at both ends of cable runs. In this way the electromagnetic shield provided by the equipment chassis and the cable screens will be optimised to reject interference. It is recommended that only high quality braided screen cables are used to avoid compromising EMC performance.

CONNECTIONS

Microphone Input

Type:	Electronically balanced
Connector:	Female XLR
Input Impedance	>1k5
Maximum Input	+20dBu

Line Input

Type:	Electronically balanced
Connector:	$^1/_4$" TRS Jack Socket
Input Impedance	>10k
Maximum Input	+20dBu

Tape Input

Type:	Electronically balanced
Connector:	$^1/_4$" TRS Jack Socket
Input Impedance	>10k
Maximum Input	+20dBu

Stereo Inputs

Type:	Electronically balanced
Connector:	$^1/_4$" TRS Jack Socket
Input Impedance	>10k
Maximum Input Level	+20dBu

Play Inputs

Type:	Electronically balanced
Connector:	Female XLR
Input Impedance	>10k
Maximum Input	+20dBu

Insert Return

Type:	Electronically balanced
Connector:	$^1/_4$" TRS Jack Socket
Input Impedance	>10k
Maximum Input Level	+20dBu

Insert Send

Type:	Ground sensing
Connector:	$^1/_4$" TRS Jack Socket
Output Impedance	<75R
Maximum Output	+20dBu

Auxiliary Output

Type:	Electronically balanced
Connector:	Male XLR
Output Impedance	<75R
Maximum Output	+26dBu

Group Output

Type:	Electronically balanced
Connector:	25 Pin D-sub
Output Impedance	<75R
Maximum Output	+26dBu

Stereo Mix Output

Type:	Electronically balanced
Connector:	Male XLR
Output Impedance	<75R
Maximum Output	+26dBu

All Monitoring Outputs

Type:	Ground sensing
Connector:	Male XLR
Output Impedance	<75R
Maximum Output	+20dBu

RECORDING WITH THE ASP8024

IN-LINE ARCHITECTURE.

ASP8024 has an In-line architecture. This means that the 'channel path' and the 'monitor path' are both included in the same physical strip.

Because the two signal paths are integrated in this way, in-line consoles have sometimes been seen as confusing. However great care has been taken in the cosmetic and ergonomic design of the ASP8024 to make the two paths easily distinguishable from one another.

Dark areas of the control surface and dark switch buttons are associated with the short fader (**SF**) or channel path while light areas of the control surface are associated with the long fader (**LF**) or monitoring path. **For the remainder of this manual the signal paths will be referred to as the LF and the SF paths.**

As a default condition the Mic/Line input feeds the SF path while the Tape return signal uses the LF path. This assignnment can be reversed using the FLIP switch. To identify which mode has been selected back lit legends indicate the signal source for each path.

THE RECORDING PROCESS

Recording is generally a two stage process, unless you are making a classical recording where very simple microphone configurations are often used to record straight down to stereo. Popular music is usually tracked first of all to get all the instruments onto some storage medium — usually a multi-track tape recorder. The second stage of the process involves returning the recorded tracks back into the console to combine them into a stereo mix. When instruments are under midi control it is not necessary to record them as they can be played live into the mix when required. If a mix has many midi controlled instruments then much of the tracking stage of the recording process can be eliminated.

BASIC TRACKING

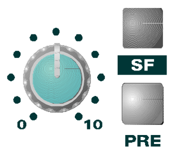
This is the first stage in the creation of a title. The starting point may well be a blank reel of tape on the multi-track recorder which is of course fully connected to the mixing console. The group outputs feed the inputs to the different tracks of the recorder while the recorder outputs are connected to the tape inputs on the console.

The FLIP switch should be in the UP position as this will ensure that the microphone or line inputs travel through the SF path to the tape recorder while the tape returns will travel through the LF path to the stereo mix bus.

A guide track is often recorded first. This could be anything from a click track to a drum kit to some very raw vocals - anything in fact that subsequent recordings can be based on - the guide track will then most likely be destroyed.

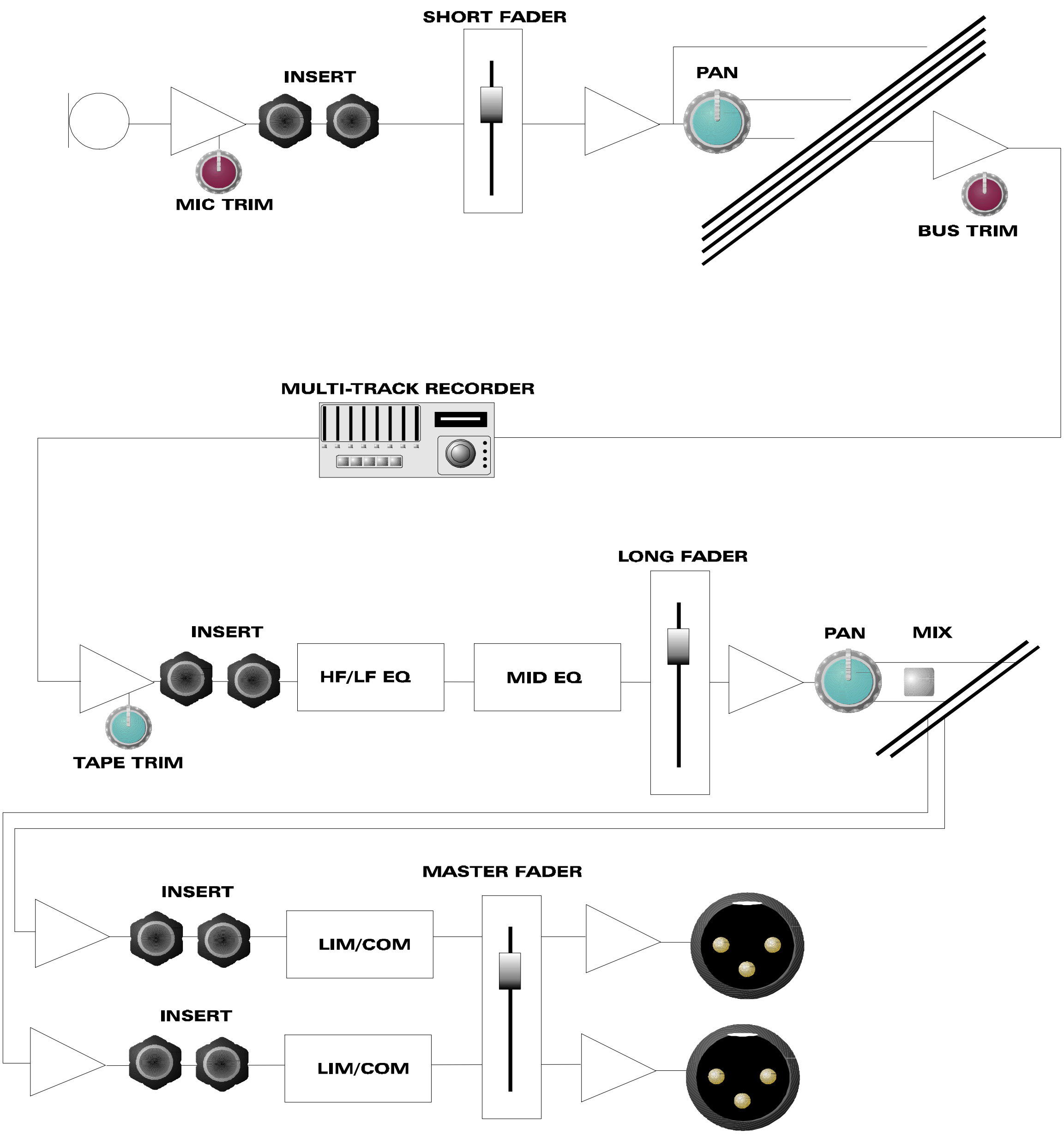
Having established a guide track it is then important to be able to hear it in order that subsequent tracks can be laid down in time with it.

Enter the auxiliaries and foldbacks !



The guide track can now be played through the LF path of the console where it can be used as a source for AUXILIARY A simply by turning up the control for AUX A on the channel that is carrying the guide track. At this time it is not really necessary to have the long fader open so switch the auxiliary to PRE and ensure that it is selected for use with the LF path. On the master section of the console AUXILIARY A can now be selected as the source for FOLDBACK A. The output of FOLDBACK A should be connected to a headphone amplifier and the performer of course needs to wear the headphones.

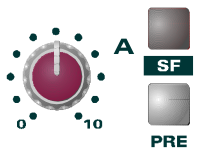
Let's assume that the performer is male, playing a guitar and the guide track is a steady drum beat previously laid down by the band's drummer. Another track on the recorder is put into REC-READY mode and the guitarist starts playing. The guitar is plugged into an amplifier and in this instance let us assume that there is a microphone placed in front of the guitar loudspeaker. The microphone is connected into the MIC input of a channel on the console which sends it through the SF path to the routing matrix.



ASP8024 Simplified Signal Flow

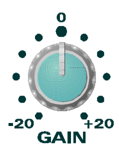
Here a decision has to be made about which track the instrument will be recorded on and this of course should coincide with the track that you are record-readying.

With the tape recorder now in record (usually a red light indicates that a track is recording) the guitar will now be recorded. Subsequent instruments can be added in a similar fashion. The recorded guitar can now be played back through the console and added to AUXILIARY A and hence to the foldback output. The next performer to record will then have the benefit of the guide track plus the guitar.



At this point a rough mix can start to be considered. Opening the long faders corresponding to tracks that are recorded will allow them to be heard on the mix output of the console (the MIX button must be pressed on the relevant channels and MIX should be selected as the control room loudspeaker source). By keeping the auxiliaries pre-fade you can have all the fun you want adjusting the stereo mix without upsetting anyone in the studio !

As the tracks are being assembled the rough mix will become more and more refined - almost starting to sound like the final title. Effect units will be starting to play their part and the auxiliary outputs will be in use feeding them. The output of the effect units will, in most cases, be brought back through the RETURNS situated in the master section of the console.



It may be of course be that the artist wants to record while listening to the effects and this can be done by turning up the Foldback 1 or 2 controls on the RETURNS. This routes the signal from the return to foldback

BASIC MIXING

As has been seen the mix has largely been created as a result of the tracking process. There may however be more instruments to be brought into the mix (some may not even require to be played since they are connected to a midi system !). These can be connected into any remaining tape inputs on the console or if all tape inputs are used or the input sensitivity of the tape input is insufficient the MIC/LINE inputs can be used.

This of course means that the signal will travel through the SF path of the console. This is no problem, however, as the SF path can be switched onto the stereo mix bus. Alternatively the FLIP switch can be used to enable the MIC/LINE input to feed the LF path if it is not in use.

For any signal using the SF path during mixdown there is no real shortage of facilities since the equaliser (or a part of it) and the auxiliaries can be switched over to the SF path.

A compressor is available on the mix output in situations where it is desirable to decrease the dynamic range of the mixed signal. The mix output also has an insert point where an external equaliser, compressor or other effect unit can be used.

INSERT POINTS

Both the SF signal path and the LF signal path have insert points. An insert point allows the internal signal path of the console to be broken into and some form of additional signal processing to be inserted. The Mic/Line signal path insert point is located immediately after the Phase Reverse circuit and just before the Flip switch. The Tape signal path insert point is located immediately after the tape input and again just before the Flip switch.

Insert points have a send and a return. The returns are fully balanced while the sends are ground sensing allowing them to be connected to balanced or unbalanced loads. If an insert point is not required and is switched out of circuit then the send still carries signal giving an additional output per insert point.

If an insert point is switched in and nothing is plugged into the RETURN jack then signal will still pass through as the jack has switching contacts which can only be broken by the insertion of a jack plug.

The insert points use TRS (Tip, Ring and Sleeve) jacks. The Tip is equivalent to pin 2 of an XLR connector while the Ring is equivalent to pin 3. The Sleeve is ground which is equivalent to pin 1 on an XLR connector.

DOING IT LIVE.....

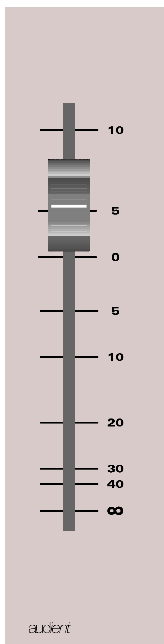
With much equipment capable of being played via midi it may be that the tracking stage of a project can be eliminated and a "live" mix created straight away. The 8024 can easily cope with this method of working since both the LF and SF signal paths can be routed to the stereo mix giving up to 72 channels to mix with.

This utilizes the electronics in the console to a high degree and almost the only part of the signal path not fully used is the routing matrix although by using the subgroups it will also be in use. Remember also that the group outputs can be used as extra effect sends either from the LF or the SF signal path.

Working this way you will want to use the SF switches to assign some resources over to the SF signal path, remembering that if it is in use in one path it cannot be used (on a module by module basis) by the other signal path.

Working in this mode it is very possible that you will want to send signals on the same module but from the different signal paths to the same effect unit.

Normally this would be a problem since the auxiliary can only be used on one or other signal path. Two auxiliaries therefore would have to be used and then combined in some way. The 8024 makes doing this very simple using the LINK switches associated with the auxiliary master controls.

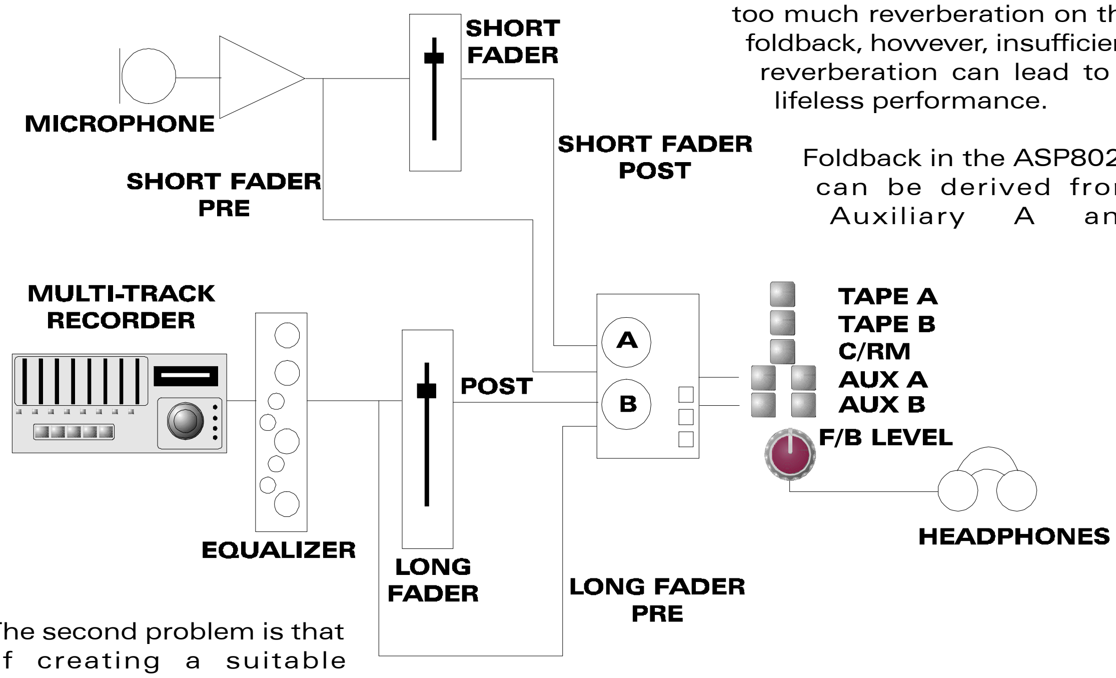


THE USE OF FOLDBACK

Foldback is the means whereby a performer can hear previously recorded material and probably other current performers in order that he/she can play along in time both with them and the pre recorded material.

Loudspeakers and headphones can be used for foldback although in most situations headphones are the usual choice. Several problems can occur if the foldback is not giving the performer what he/she expects to hear leading to timing and pitch variations.

The first problem is of course the number of available foldback mixes. If the number of performers exceeds this then an element of compromise is already introduced.



The second problem is that of creating a suitable foldback mix. Listening on headphones is a very unnatural experience for most people. Checking the foldback mix on the control room loudspeakers or even on headphones is unlikely to sound the same as it is perceived by the performer listening in the studio.

Should the foldback be mono or stereo ? Generally it has been found easier to follow a beat that is in mono. If a click track is being followed than it may pay to vary the click sound as the performer's hearing adjusts to the click and desensitizes to it. Increasing the level of the click is of course an option but this will quickly lead to more desensitization and the risk of requiring to increase the level yet again.

The level of the performer relative to the remaining foldback is a critical point.

If timing is a problem it may be that the performer is hearing too much of himself and not enough of the rest of the performers. A solution here is to remove one of the ear pieces so the performer can retain a feel for his own instrument while getting more of the others.

For vocalists the pitching may be a little sharp if there is insufficient of themselves in the foldback to be able to judge pitch. The converse is also true where if there is too much of the vocalist in the foldback then pitching may be flat. Pitching is usually easier with a stereo foldback mix.

If pitching is erratic it may be that there is too much reverberation on the foldback, however, insufficient reverberation can lead to a lifeless performance.

Foldback in the ASP8024 can be derived from Auxiliary A and Auxiliary B.

It is also possible to select the CONTROL ROOM SOURCE or the EXTERNAL INPUTS to the monitor system as foldback sources. Additionally the stereo returns can be sent to the foldback system allowing effects to be heard by the performers.

Auxiliaries A and B are normally derived from the LF path of the console but can be switched to take signal from the SF path. The signal is normally post fader but can be selected to be the pre-fade signal. The foldback outputs are line level outputs and require to be connected to a suitable amplifier. They are not intended to directly drive headphones or loudspeakers.

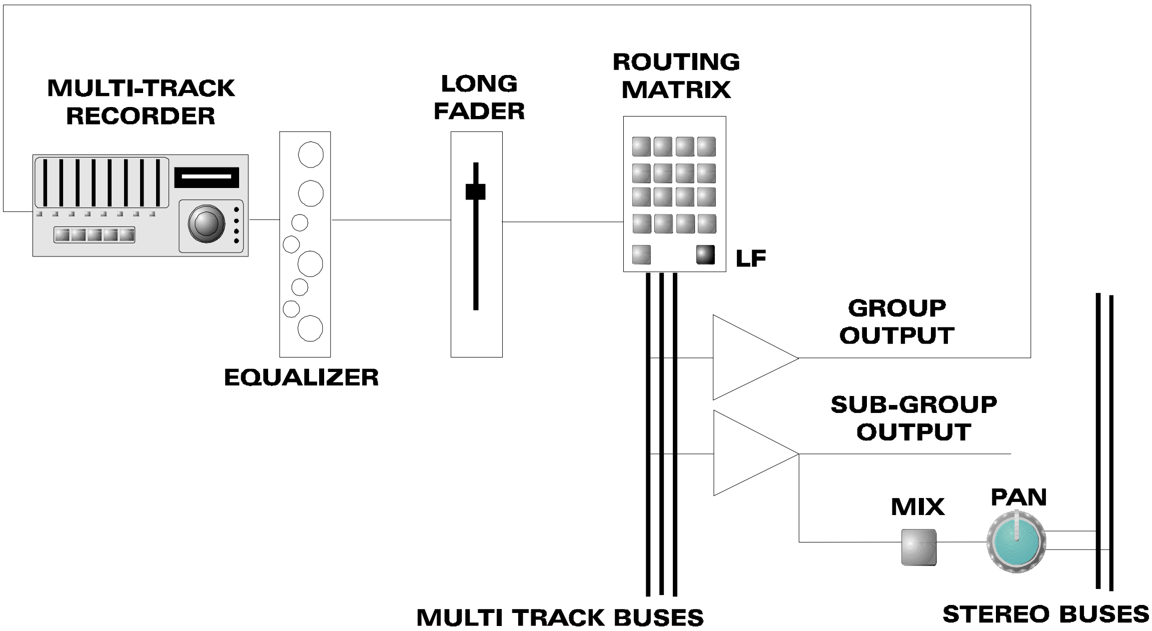
GROUPS AND SUB GROUPS

It is the group outputs of the console that are connected to the multi-track recorder inputs. Consequently to record anything it has to be routed to a group output.

This is done by using the routing buttons at the top of the channel strip where there are 12 buttons plus a shift button allowing routing to all 24 group outputs. The group is reached indirectly through the group mix bus. The mix bus allows other signals to be routed to the same group output giving the possibility of a track carrying only one instrument or a combination of instruments.

through the console also means that the equalizers can be used to modify the signals. The mixed signal can of course be monitored through the stereo bus.

The auxiliaries can also be used to send, for example, to a reverberation device. The output of the reverberation unit can be brought back through a stereo input to be used for foldback and can also be recorded either as a part of the “bounced track” or on a separate track.



The disadvantage of combining instruments at the recording stage is of course that the mix between them is then fixed and cannot be changed at a later date.

A better idea (but only if you are short of tracks) is to record the instruments separately and then bounce them down to another track. This is achieved by routing the LF path up to the routing matrix on all the channels to be bounced and selecting a track for the mixed signal to be recorded on (it must be the same for all signals). The tape is then played and the mixed signal re-recorded on a new track. Be careful when bouncing to an adjacent track on an analogue recorder, crosstalk in the record head may cause feedback.

This can be done until the mixed track is deemed acceptable. Bringing the signal

When mixing down, especially if automation is not available, it can be useful to sub mix certain tracks, such as the backing vocals, together. This is possible by again assigning the LF signal up to the routing matrix. The first 8 groups are also automatically assigned to 8 subgroups, thus group 1 also routes to sub group 1, group 2 also routes to sub group 2 and so on.

The sub group outputs can be used directly but in this situation it is more useful to assign the sub groups to the stereo mix by pressing the MIX button. Any signals requiring to be sub-grouped can now be routed to a subgroup and the level within that subgroup is determined by the long fader. The overall subgroup level is determined by its fader, located in the centre section of the console. It is possible to have 8 subgroups to simplify and reduce the number of faders requiring adjustment during a mix down session.

OVERDUBBING

Overdubbing is really a combination of the tracking and mixing modes. In fact most tracking that is done will be done as overdubs and the only pure tracking may in fact be the laying down of the guide track when there is nothing else on the tape.

Both the LF path and the SF path of the console are used in overdub mode. The LF path is used to bring the tape playback signals both to the stereo bus and to the auxiliaries from where it can be sent to the performers through the foldback system.

The SF path is the recording path and is used to take input from the studio and send it to a track on the multi-track recorder.

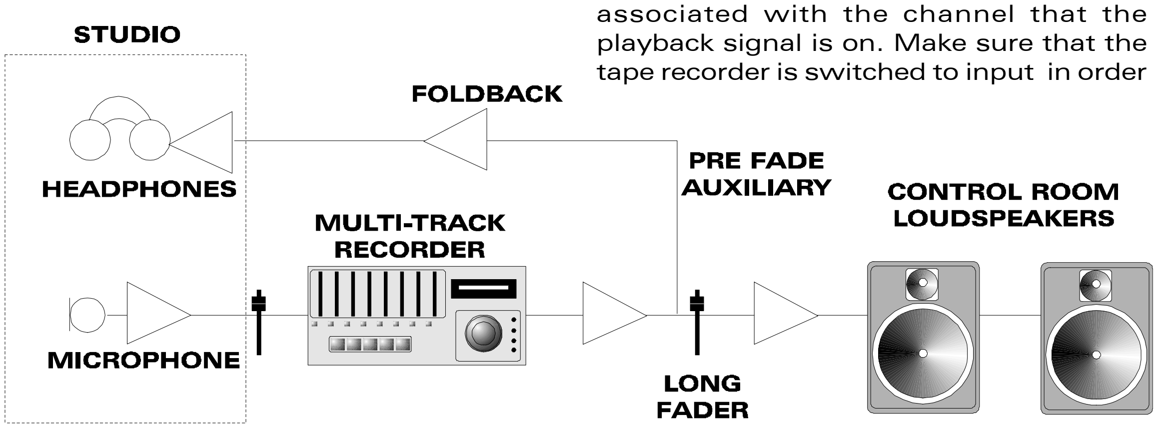
To use foldback the signal from tape should be assigned to auxiliary A or B. The auxiliary should be assigned to the LF path and could be either pre or post fade.

short fader. The input meter will help determine this and the level at this point can be controlled by the input sensitivity control.

The signal then travels through the short fader where the level to tape may be controlled and thence to the routing matrix where it can be assigned to a track (or tracks) on the recorder.

The track to which the overdub is assigned should be put in REC-READY mode with the remaining tracks in SAFE mode. Thus when the RECORD button is pressed only the track which is rec-readied will actually go into record. Be very careful not to overwrite a previously recorded track. The use of track sheets, where a list of all the instruments on each track is kept, is recommended.

The signal level being sent to the tape recorder can be seen on either the tape recorder meters or on the large meter associated with the channel that the playback signal is on. Make sure that the tape recorder is switched to input in order



If pre fade then you will be able to play with the stereo mix without any effect on the foldback. A post fade setting may be preferred if changes to the mix are to be reflected in the foldback signal.

The recording path uses the mic/line input and this will be connected to a microphone, D.I. box or other signal source within the studio. The input signal can be checked on the small meter available on the input pod or by using the meter reverse switch the long meter can be used. This does not indicate the level being sent to the tape recorder, only the signal level immediately after the insert point in the SF signal path.

The level of course is important and it must not be too high or low even before the

that the recorder input signal is returned to the console for viewing on the meters. Most machines incorporate an auto-input function allowing the rec-readied tracks to automatically switch to input. Tracks which are SAFE will remain in TAPE mode.

You should consult your tape recorder manual for more information regarding this.

Levels can be difficult to set and there may be large differences between the meters on the recorder (especially if it is analogue) and those of the console.

The section on Metering later in this manual, gives more information about the meters used in the ASP8024.

PATH SWAPPING

The simplified block diagram on this page shows the short and long fader paths. The FLIP switch is at the input of both paths and determines whether the MIC/LINE or the TAPE input is used for a particular path.

The normal position for all switches on the console is UP and with the FLIP switch in this position the TAPE signal is routed through the LF path as shown.

By studying the diagram it can be seen that the MIC/LINE input is routed through the SF path but that when the FLIP switch is pressed this situation is reversed and the TAPE signal is now routed through the SF path.

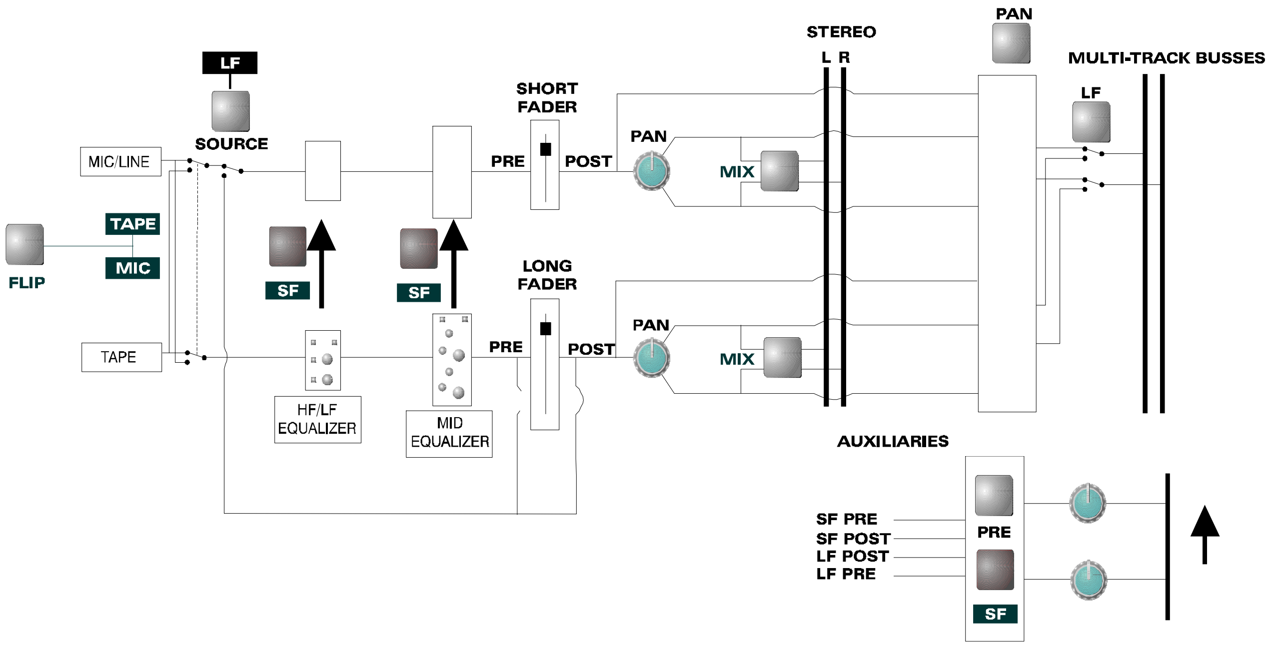
There are switches labelled SF associated with both the HF/LF equaliser and the MID equaliser. Again the normal position of these switches is UP, placing the equalisers in the LF path.

By pressing the SF switches the equaliser sections can be moved over to the SF path.

Thus it is possible to have the entire equaliser in either the LF or SF path or one section of the equaliser in each path. This situation could be useful when both the LF and SF paths are used for mixdown.

If the multi-track recordings are to be made with equalisation then the equalizers should be placed in the SF path. Note in this situation that you will hear the effect of the equalisation and it will be recorded. When the equaliser is in the LF path you will hear the changes but they will not be recorded on the multi-track recorder. They will only be recorded if a stereo recording is made, as will be the case when mixing the title.

It is possible the source the auxiliaries from the SF path by pressing the SF switch associated with each pair of auxiliaries. The auxiliaries can further be selected to be either pre or post fader for whichever path they are in.



TRACK BOUNCING

Track bouncing is used to mix a limited number of tracks together and rerecord them. This may be to free up some tracks on the tape recorder so that additional material may be recorded or it may be so that the final mix is simplified by having certain groups of instruments already premixed. This could also be done, during mixdown, by using the sub group facility.

Signal coming off tape is normally sent through the LF path. To rerecord it is necessary to route the signal to the routing matrix to assign the signal to the designated recording track.

The FLIP switch could obviously be used to route the playback signal through the SF path and hence back to the tape machine but this would necessitate replicating panning and EQ in the SF path. However, it is preferable to use the LF button located by the routing matrix which will select the output of the long fader to the routing matrix. This will preserve any panning and EQ that has been set up. Ensure that all the tracks to be bounced are unrouted from the mix. The end result can be heard by assigning the destination track's LF path to the mix.

SOLO

Pressing a solo button on a channel with either PFL or AFL selected allows either the PFL or AFL signal for that channel to be heard on the monitors and viewed on the stereo output meters.

Associated with the solo switch is the SOLO-IN-FRONT control and this allows the relative level of the solo'd signal and the stereo mix to be adjusted. It is thus possible to hear a channel in isolation or with some amount of the mix behind it.

Solo-in-Place is an extension of the AFL and PFL facilities. If SIP is selected on the master module the signal on the main stereo bus will be replaced by the AFL (Post Pan) signal of the soloed channel. This is the equivalent of cutting all other channels except the one you want to check, but is achieved by just a single button push.

Note that if you are recording when Solo in Place is used this will be recorded !

For this reason the SIP switch is illuminated alerting users to possible danger. The most likely use for SIP will be just before a mix when equalization is being set up. It is often easier to adjust the equalization on a solo'd channel but it is essential to check how it sounds when in the mix.

METERING

Metering is extremely important as it can determine whether or not a signal is too low in level, in which case it may be noisy, or too high in level in which case it may suffer distortion. The aim of the meters is to assist in setting the signal level between the two extremes of noise and distortion.

A signal level that is too high clips meaning that the smooth waveform abruptly changes when the electronics runs out of headroom. Normally a visual indication of impending clipping is given before the audible effects become apparent.

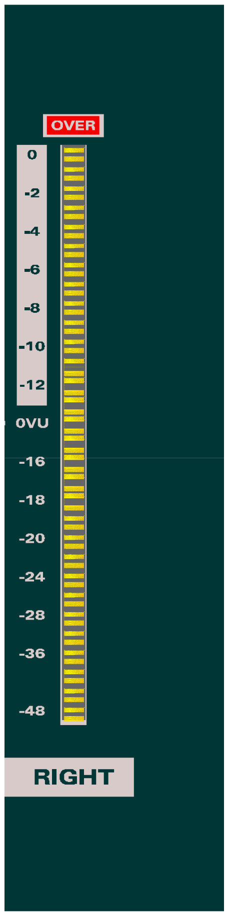
At the other extreme, if there is insufficient signal, any noise present will be amplified along with the signal when it is eventually brought up to the correct level. There are no indicators warning you of this condition - only the fact that the signal is not "peaking" to any extent on the meter.

To aid this situation the audient 8024 has input and output meters. Tape Input levels can be monitored on a 20 segment meter, while microphone input levels can be monitored on a 3 segment meter. Provision is made for the function of these meters to be reversed, as, particularly when tracking, it may be desirable to observe the input signal on the large meter.

On most analogue consoles it can be a very trying task to try and associate the VU meter readings with those on a digital tape recorder. The ASP8024 peak reading meters make this task simpler by effectively replicating the digital meters on the recorder. Since the meters are peak reading they give a true indication of the signal level allowing you to use the full dynamic range of the electronics.

The meters are calibrated with 0dBFS at the top of the scale, all other levels being negative with respect to this.

OdB is calibrated for a signal level of +18dBu and the 0VU mark for +4dBu. In addition there is an **OVER** indicator which illuminates at +20dBu. The meters are peak reading to give a response much more like the readings on a digital recorder.



Meters are provided for :-

Main Stereo Output

Subgroup Outputs

Tape Return Inputs

MIC/LINE Inputs

MORE ON CONNECTORS

JACK SOCKETS

Jack sockets are a very common form of connector and are used extensively in the 8024 console. There is more to them, however, than merely connecting with the Tip, Ring and Sleeve of the mating jack plug.

The switching contacts allow the jacks sockets to be used to pass a signal through while no mating plug is inserted but break this signal and accept the signal from the plug when it is inserted. This is known as "normalling" and the signal connected through the switching contacts is known as the "normalled" signal.

The prime use of this function within the console is on the insert points. If this facility did not exist and the insert point was switched in, nothing could pass through the signal path unless an external unit was connected between the insert send and the insert return jacks. If the insert return jack has its normal contacts wired to the signal being sent to the insert send, then, if no jack is inserted in to the return socket the send signal will pass through the contacts and appear on the output of the jack.

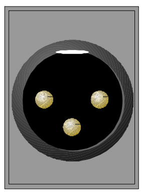
This allows the insert point to be left switched into circuit at all times with the only consideration being that the signal passes through one more stage of electronics than it otherwise would.

The insert sends are ground sensing outputs enabling them to connect to balanced and unbalanced loads with equal ease. When connected to an unbalanced load any ground noise from the destination is also added to the signal making it a common mode signal which is rejected.

XLRS

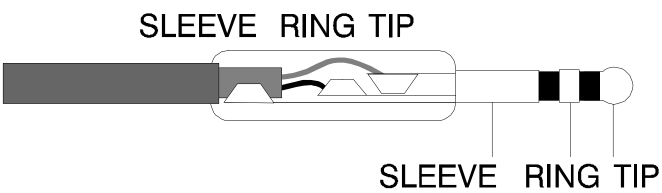
XLR connectors are the connectors found on many items of professional audio equipment. XLR stands for "extra low resistance" as there is a large contact area associated with this connector. Male connectors are used as outputs while the female connectors are used as inputs. There are 3 pins on each connector with pin 1 always being used as the earth or ground pin to which the cable shield should be connected.

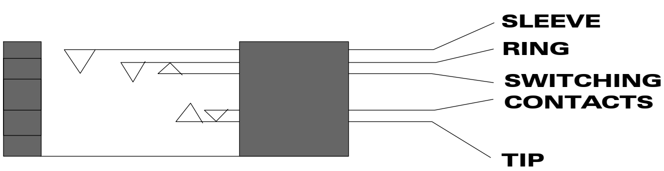
The balanced signal is carried on pins 2 and 3, Pin 2 is normally the Hot pin while pin 3 is Cold pin. In the USA generally the reverse is true and pin 3 is the Hot pin. With balanced circuits throughout this makes little or no difference but beware if unbalanced inputs and outputs exist as this may be the cause of a polarity reversal, or signal short circuits.



MALE XLR

FEMALE XLR





GLOSSARY

AFL
This allows after fade (post fade) signals to be heard on the monitors and viewed on the main stereo meters.

AUXILIARIES
Sometimes known as auxiliary sends these are used as secondary mix buses. The mixes created on these buses are then used to feed effect units or are fed back to the performers as a Foldback feed. Every channel has access to the auxiliary mixes and the contribution of any channel can be varied by using the appropriate auxiliary level control.

AUXILIARY MASTER
Associated with every auxiliary there is an auxiliary master control to give overall level control rather than having to adjust the contribution from every channel. Auxiliaries can be switched pre or post fader and be allocated to either the SF or LF signal paths. A further switch assigns the controls to a different pair of auxiliary buses, reducing the number of controls needed for a given number of auxiliaries.

BOUNCING
Bouncing (or track bouncing to give it its full name) is the process of moving one or more tracks from their existing position(s) to a new track. This is most often done in order to free up more tracks for recording.

BUS MASTER
Inputs are assigned to group outputs through a bus enabling one or many inputs to be assigned to the same bus. The group outputs usually correspond to inputs on the multi-track recorder. Each bus then has a mixing amplifier whose gain can be controlled by the Bus Master Trim. This allows the level to a multi-track input to be raised or lowered without having to adjust the individual level of each channel routed to that track. The faders can still be used to adjust the relative levels of channels.

CUT
The cut or mute control is used to silence (mute) a signal path when it is not in use. This removes the noise contribution from that source leading to a quieter mix. There are cut switches in both the SF and LF signal paths.

DIM
This allows the control room loudspeaker levels to be reduced by a preset amount. Dim will be brought into action automatically when talkback is used preventing howl round.

EQUALISER
Equalisers are what would be referred to as tone controls on consumer equipment. Equalisers are divided into a number of bands - 4 in this case. There is scope to adjust high and low frequencies and two bands of middle (mid) frequencies. The high and low frequency sections are shelving and the turnover frequency is switchable. The middle frequency sections are peaking and the frequency of the peak (or dip) is adjustable. It is also possible to alter the Q of the mid sections with a pot, making the Q continuously variable between two values. In line consoles often have the facility for the equaliser to be split such that it can be used partly in the channel path and partly in the monitor path. The SF switches on the 8024 equalisers allow the HF/LF and MID equalisers to be independently switched into the SF signal path.

FLIP
Flip allows the inputs to the signal paths to be swapped. Normally the LF path will carry the TAPE input, however, with FLIP pressed it will carry the MIC/LINE input while the TAPE input will travel through the SF path.

FOLDBACK
Foldback is a mix that is returned to the performers in the studio in order that they can play in time with what is already recorded. It could simply be the console stereo output although more usually it is taken from a pair of auxiliary buses allowing a different mix to be created. Talkback may also be included on the foldback outputs enabling communication with the artists.

IN LINE
This refers to a type of console which contains two signal paths within a module. The channel signal path is used to feed a multi-track tape recorder while the monitor path is used to carry the output of the multi-track recorder through to the stereo mix bus. In line consoles can be more compact than split consoles or can carry more channels for a given size. The possibility of switching signals between the two paths and of sharing facilities between the paths makes the in line concept a very attractive one.

INSERT POINTS
Insert points allow the signal path to be broken allowing the insertion of some signal processing device. The device inserted is then in series with the signal path. When not required the device can either be switched out using the INSERT switch or unplugged from the console.

LF
The Long fader is normally used to feed the mix. The lighter areas of the channel strips are areas used for the LF signal path. There may be a dark switch labelled SF allowing that facility to be switched into the SF or short fader path.

LINE
The line input is a high level, high input impedance input intended for high level sources such as the outputs of a multi-track tape recorder, sampler etc.

MIC
The microhone input is a low level, low impedance input intended for use by low output devices such as microphones. This contrasts with the line input which is intended for use by equipment with high output levels, a tape recorder line output for example.

MIX
This allows signal to be routed to the stereo mix bus which is the main output of the console. This routing can be applied to both the SF and LF signal paths and is particularly useful during mix down when as many inputs as possible are often required.

PAN
Short for panoramic potentiometer this control places a mono source signal onto the stereo bus. The proportion of signal fed to the left and right buses is variable (using the pan control) and alters the spatial position of an instrument within the mix. Thus a number of channels can all be panned to different spatial positions. Generally low frequency instruments such as kick drums are panned centrally as they are omnidirectional and for a given SPL the speakers are being driven at a lower level leading to less distortion. Signals can also be panned across odd and even group outputs allowing them to recorded in stereo on the multi-track recorder.

PFL
This allows pre-fade signals to be heard on the monitors and viewed on the main stereo meters of the console.

Q
Q is an indication of the frequency range or bandwidth over which a peaking equaliser will be effective. Low Qs affect a wide range of frequencies while high Qs affect a much narrower range of frequencies. It thus allows an equalisation adjustment to be targeted to maximise the effect where required while at the same time minimising changes where they are not wanted. Generally high Qs sound less pleasant than low Qs.

ROUTING

This is the process of selecting to which group output of the console the signal should be routed. Routing can be to multiple tracks and if an odd/even combination is selected then panning can be used to record a stereo signal onto the multi-track recorder.

SF
The Short fader is normally used to feed the inputs to a multi-track tape recorder. It therefore controls the recorded level of the signal. When SF appears by a control or group of controls it means that these functions can be switched into the short fader (or channel fader) path.

SHIFT
This allows the number of routing switches to be reduced by doubling the function of each switch. With Shift unpressed routing is possible to tracks 1 through 12. With Shift pressed routing to tracks 13 through 24 is possible.

SOLO IN PLACE
This is a method of previewing the signal in a channel and works by cutting all the signals feeding the stereo bus other than the one(s) being solo'd. This is a destructive process and does affect the stereo or mix output of the console.

STEREO BUS
This is usually the main bus in the console and provides the output to whatever stereo recording device is in use. The stereo output is also used as the main monitor source allowing the output of the multi-track to be heard and the balance of the individual tracks in the mix to be adjusted.

TALKBACK
Talkback is a means of communication from the mixing console to the performer.

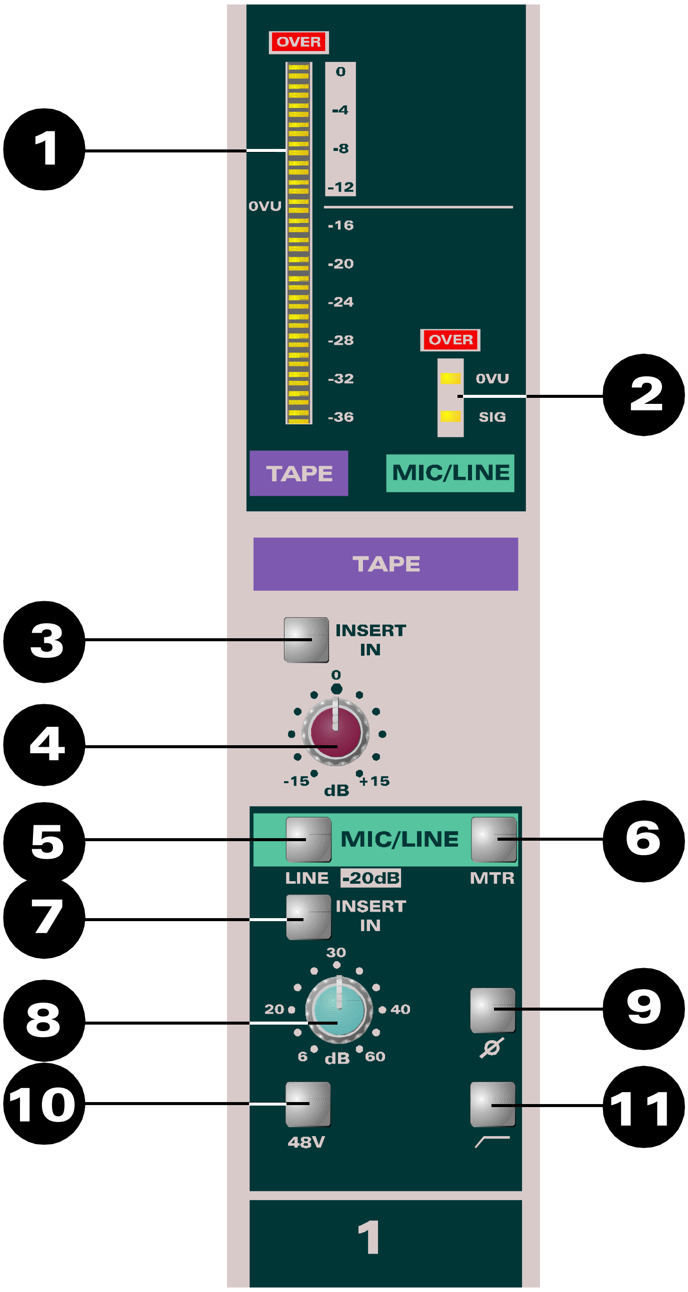
CONSOLE FUNCTIONS

Now that an understanding of the use of the console has been gained by example it is time to list the functions on the modules. The modules have been sectioned in order to show them at a reasonable size and the controls identified by number. A brief functional description of the controls is given below the drawing while the side bar gives a broader overview of the functions.

This section of the manual can be treated as a reference section when looking for a specific control.

ASP 8024

audient



1 20 Segment meter

2 3 Segment meter

3 **INSERT IN** places an insert point in the TAPE path

4 Tape Input **TRIM** control

5 **MIC/LINE** switch - press down to select the line input.

6 **MTR** - press to show the mic/line input on the large meter and the tape input on the small meter.

7 **INSERT IN** - places an insert point in the MIC/LINE path.

8 Mic/Line **GAIN** Control

9 **Ø** Polarity (Phase) Reverse Switch

10 **48V** Phantom Power Switch. Turn the loudspeakers down before switching this on or off !

11 High Pass Filter Switch. A high pass filter can be used to get rid of any unwanted low frequencies that may be present such as air conditioning rumble.

INPUT POD

The input pod is the gateway to the remainder of the signal processing of the console. There are three inputs, a mutually exclusive microphone and line input, and a tape input.

Note the different colouring used to identify the different signal paths of the pod. Anything on a light background is associated with the LF path while anything on a dark background is associated with the SF path.

The mic/line input normally sends signal to the SF or short fader signal path of the console while the tape input normally sends signal through the LF or long fader path of the console unless this is reversed by the FLIP switch.

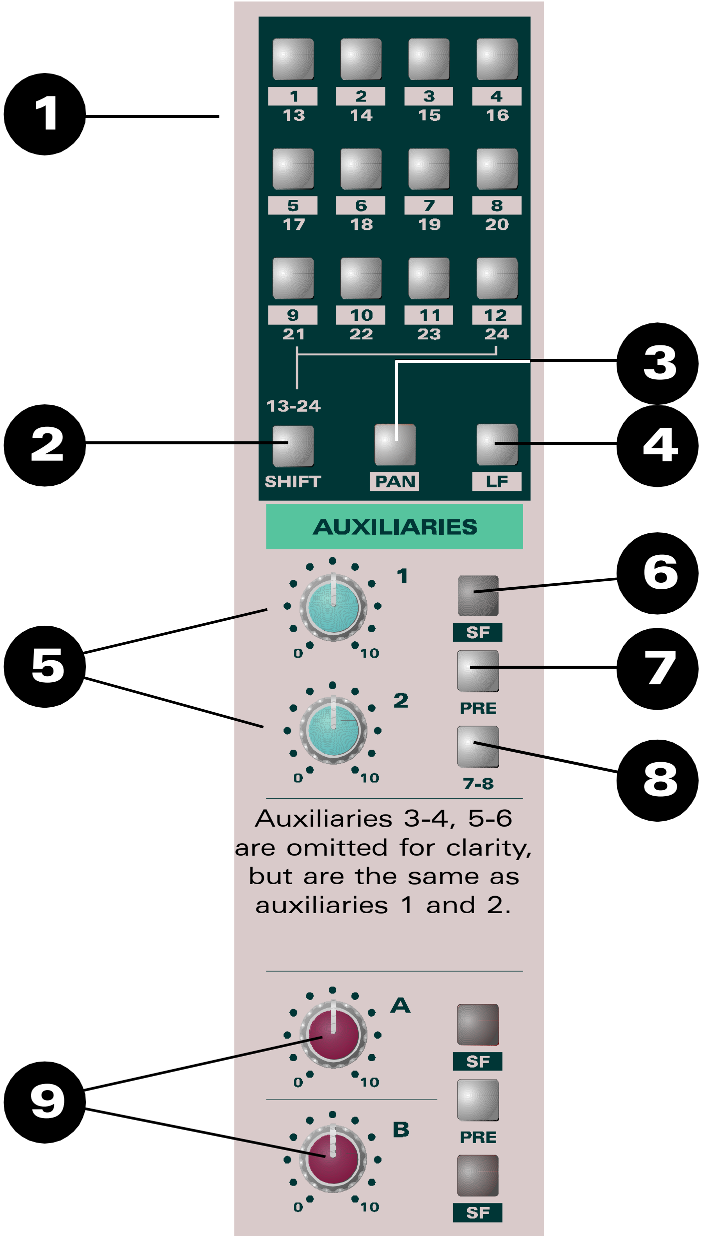
There are two meters associated with the inputs, a 20 segment 0dBFS peak reading meter showing the tape input signal and a 3 segment peak reading meter gives an indication of the mic/line level.

Both the MIC/LINE and the TAPE inputs have insert points which can be switched in and out of circuit.

The Mic Input has a gain control range of +6dB to +60dB and the Line Input has a range of -14dB to +20dB.Switches allow for Phantom Power, Polarity Reversal and High Pass filtering.

The TAPE input has a trim control with a range of ±15dB.

There is a back illuminated number at the bottom of the pod for channel identification.



1 **ROUTING SWITCHES** select the group outputs that the SF signal is sent to. The first eight switches also route to the 8 subgroups

2 **SHIFT** gives access to group outputs 13 - 24.

3 **PAN** allows the signal to be panned across odd and even groups.

4 **LF** replaces the SF signal being sent to the routing switches with the signal from the LF signal path.

5 **LEVEL** adjusts the level sent to an auxiliary output.

6 **SF** allows the signal feeding the auxiliary to be taken from the SF signal path.

7 **PRE** allows the auxiliary signal to be taken pre fader instead of the normal situation where it is taken post fader.

8 **7-8** allows the signal to be routed to auxiliaries 7-8 instead of 1-2.

9 **AUXILIARIES A** and **B** are identical in operation to the others but can be individually switched into the SF path.

ROUTING and AUXILIARIES

The routing section takes the signal from the SF path and routes it to the group outputs which in turn are usually connected to the inputs of a multi-track recorder. Groups 1 to 8 also have a parallel path and feed the 8 sub groups. These can be used to pre mix channels together for final mix down or as sends to an 8 track recorder.

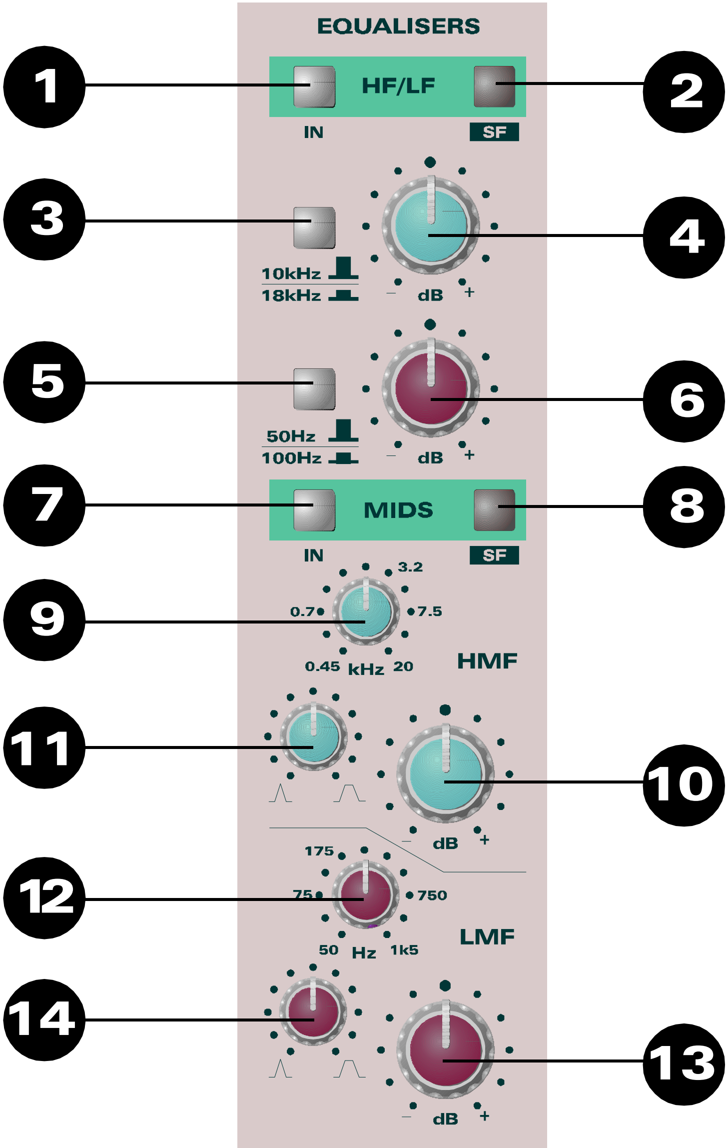
There are 24 group outputs accessed by 12 routing buttons and a SHIFT button.

Routing can be to multiple outputs and if odd and even outputs are selected panning can be used to place the signal within a stereo image.

The LF switch moves the routing from the SF path over to the LF path and can be useful for track bouncing. If PAN is down and routing is again selected for odd and even groups then the post fade post pan LF signal is sent to the group outputs selected.

There are 14 auxiliary outputs although the number of panel controls does not reflect this. Again a switch is used to assign the controls between two pairs of auxiliaries. Auxiliaries 1 and 2 can be switched for use as 7 and 8 for example. The Auxiliaries can be switched to the SF path.

Auxiliaries A and B work in the same manner as the other auxiliaries but are intended mainly for use as sends to the FOLDBACK system.



1 **HF/LF** Section. This places the HF/LF equaliser in circuit.

2 **SF** places the HF/LF equaliser in the SF signal path where it can be used to treat a signal before it is recorded.

3 **10kHz/18kHz** allows the frequency of the HF equaliser to be selected.

4 **HF boost/cut** control.

5 **50Hz/100Hz** allows the frequency of the LF equaliser to be selected.

6 **LF boost/cut** control.

7 **MIDS** Section. This places the MIDS equaliser in circuit.

8 **SF** places the MIDS equaliser in the SF signal path where it can be used to treat a signal before being recorded.

9 This controls the centre **FREQUENCY** of the high mid equaliser.

10 **HIGH MID boost/cut** control.

11 **HIGH MID Q** control.

12 This controls the centre **FREQUENCY** of the low mid equaliser.

13 **LOW MID boost/cut** control.

14 **LOW MID Q** control.

EQUALISER

Note the light background indicating that the equaliser is normally associated with the LF signal path.

The equaliser is split into two sections - one for high and low frequencies (HF/LF) and the other for middle frequencies (MIDS). Both sections can be switched in and out independently and switched into the SF signal path independently.

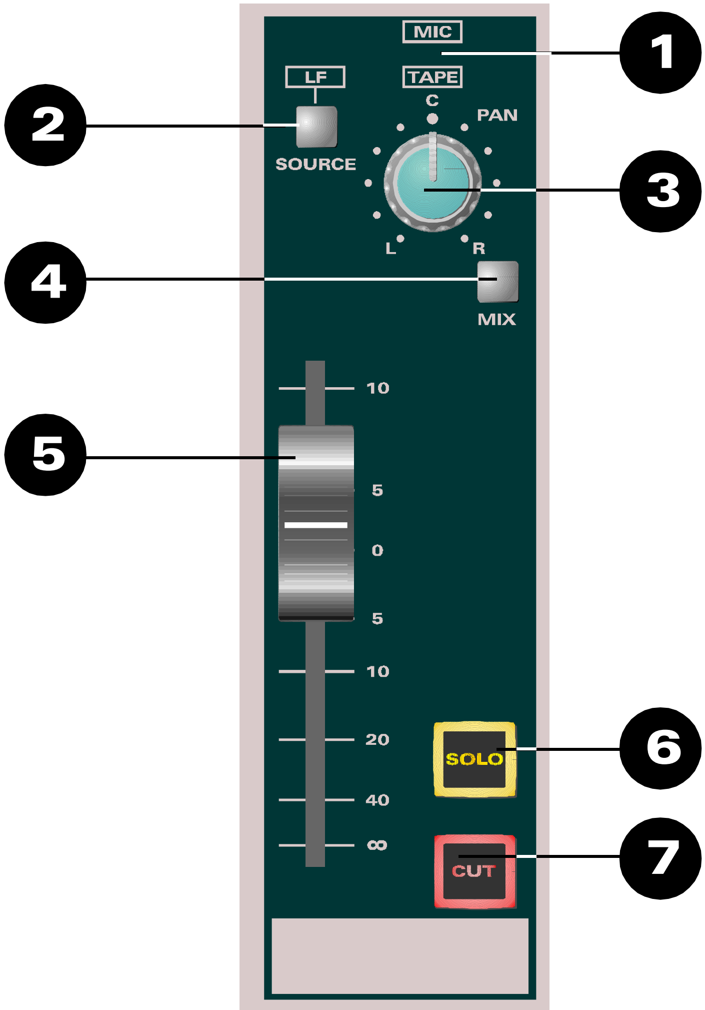
The HF section has a boost/cut range of 15dB. The frequency can be switched - 10kHz with the switch up and 18kHz with the switch depressed.

The LF section has a boost/cut range of 15dB and a shelving characteristic. The frequency can be switched - 50Hz with the switch up and 100Hz with the switch depressed.

The high mid section (HMF) has three controls. The boost/cut range is 15dB and the frequency of operation can be varied from 450Hz to 20kHz. A Q control allows for a very sharp and narrow characteristic or for a more gentle characteristic covering a wider range of frequencies.

The low mid section (LMF) has three controls. The boost/cut range is 15dB and a shelving characteristic. The frequency of operation can be varied from 50Hz to 1.5kHz. A Q control allows for a very sharp and narrow characteristic or for a more gentle characteristic covering a wider range of frequencies.

The actual Q range is between 3.8 (0.4 Octave) and 0.65 (2 Octaves).



1 MIC/TAPE. These back lit indicators show whether the MIC/LINE or the TAPE input is selected to the SF path. Only one will be illuminated and it can be changed by using the FLIP switch.

2 SOURCE selects the post fade LF signal as the input to the SF signal path, over-riding the MIC or TAPE selection from the Flip switch. By altering a link on the circuit board this signal can be made pre fade.The LF legend will illuminate when the switch is pressed and the MIC/TAPE indicator will blank.

3 This is the PAN control for the SF signal enabling it to be panned across odd and even groups.

4 MIX routes the SF signal to the stereo mix bus. It is good practice to unroute any channels which are not needed. This will reduce mix amp noise.

5 This is the SHORT FADER which controls the level of the SF signal.

6 SOLO allows the SF signal to be heard on the monitors and viewed on the master meters. If Solo In Place is selected it will replace the console output.

7 CUT allows the SF signal path to be muted. This may help to reduce noise in a mix if a channel is not in use for a period of time.

SHORT FADER (SF)

There are 3 back illuminated indicators showing the selected input to the SF path. This can be changed between Mic/Line and Tape using the FLIP switch.

LF Source allows the source for the SF path to be taken from the LF path. This could be used during mixdown to send the LF signal through the SF path up to the routing matrix where the group outputs can be used as additional effect sends. Normally this signal is derived after the long fader (POST LF) but it can be made PRE LF by changing an internal link on the circuit board.

The source switching occurs before the equaliser so it is possible to equalise the SF signal which has been taken from the LF path by switching either one or both equaliser sections into the SF path. (If this is done then the equaliser section is no longer available in the LF path).

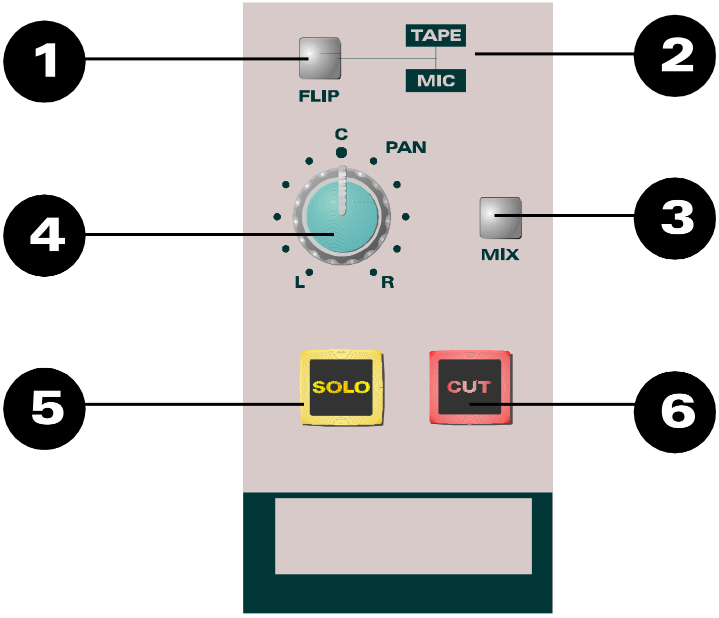
The SF PAN control allows panning across the group outputs when pan is selected on the routing section of the strip.

MIX allows the SF signal to be routed to the stereo mix bus and allows the SF path to be used as an additional input during mixdown.

The short fader is designed for use with 10dB of gain in hand allowing the signal to be boosted or reduced in level if required.

The SOLO switch allows the channel to be auditioned through the AFL/ PFL or Solo in Place facilities.

CUT allows the SF signal path to be muted.



1 FLIP swaps the inputs between the LF and SF paths.

2 These back lit indicators show whether the MIC/ LINE or the TAPE input is selected to the LF path. Only one will be illuminated and it can be changed by using the FLIP switch.

3 MIX allows the LF signal to be routed to the stereo mix bus. Normally this switch should be pressed but it is good practice to unroute any channels which are not needed. this will help to reduce mix amp noise.

4 This is the PAN control for the LF signal enabling it to be panned across the stereo bus

5 SOLO allows the LF signal to be heard on the monitors and viewed on the master meters or, if Solo in Place is selected, it will replace the console output.

6 CUT allows the LF signal path to be muted. This may help to reduce the noise in a mix when a channel is not in use for a period of time.

FLIP and PAN

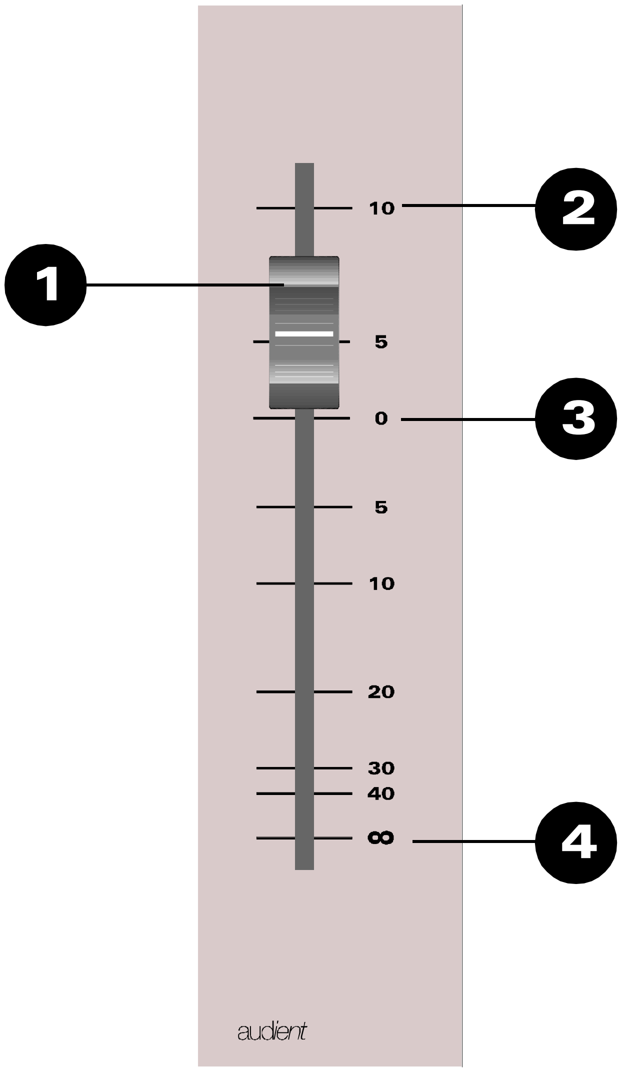
The FLIP switch allows the MIC/LINE input and the TAPE input to be reversed. Normally the TAPE input feeds the LF signal path - with FLIP pressed it will feed the SF signal path. The MIC/LINE input normally feeds the SF signal path and with FLIP pressed it will feed the LF signal path. Illuminated indicators in each section show which input is selected to the LF and SF paths.

The PAN control pans the LF signal across the stereo mix bus and the MIX switch assigns the LF signal to the stereo bus.

The SOLO switch allows the long fader signal to be auditioned through the AFL/ PFL or Solo in Place facilities.

CUT allows the LF signal path to be muted.

Page 33 gives more detail regarding the long fader used to control the level of the LF signal path.



1 This is the **LONG FADER** which controls the level of the LF signal.

2 This is the **+10dB** mark. The fader is fully open at this point and introducing 10dB of gain into the signal path.

3 This is the **0dB** mark. This is the normal operation position for the fader.

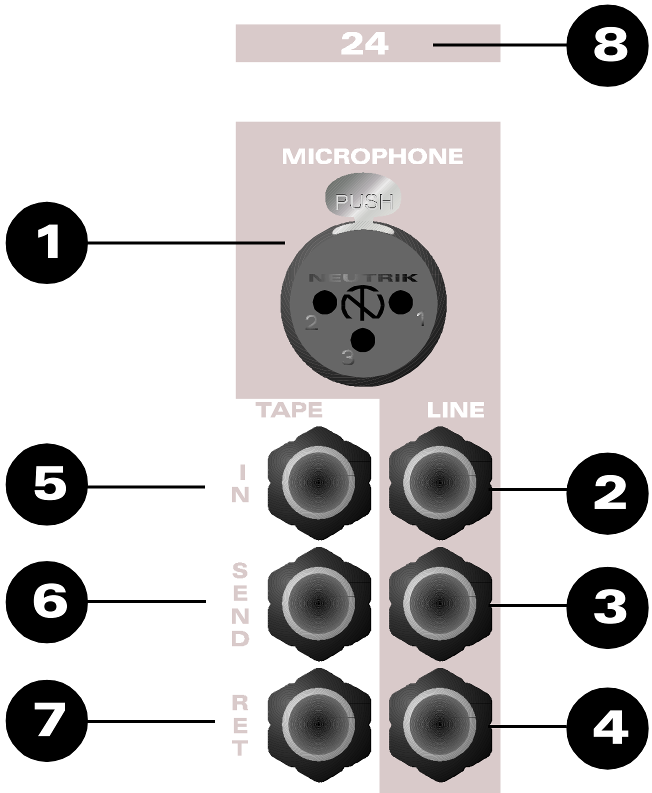
4 With the fader here the signal path is closed and no signal will pass.

LONG FADER (LF)

Located close to the operator this fader operates on the LF signal path and is therefore mainly used for creating the monitor mix and the final stereo mix for the title.

The fader is expected to operate around the 0dB mark with 10dB of gain in hand allowing the signal to be increased or decreased in level.

When level setting start with the fader in this position then adjust the input sensitivity control for the correct level to optimise the gain structure.



CONNECTOR PANEL

The rear mounted connector panel is where the input, output and insert point connectors are located.

The microphone input uses an XLR connector while the line input, tape input and the insert sends and returns use Tip, Ring and Sleeve jacks.

1 **MICROPHONE INPUT.** Microphones or other low levels inputs can be connected to this input.

2 **LINE INPUT.** This input can selected in place of the microphone input. Like the Tape input it is suitable for high level line sources.

3 **MIC/LINE INSERT SEND .** This is the insert point send output for the microphone and line input. Signal is always present here and can be used as an additional output. Only the Mic/ Line signal will appear here and it is not affected by the Flip switch.

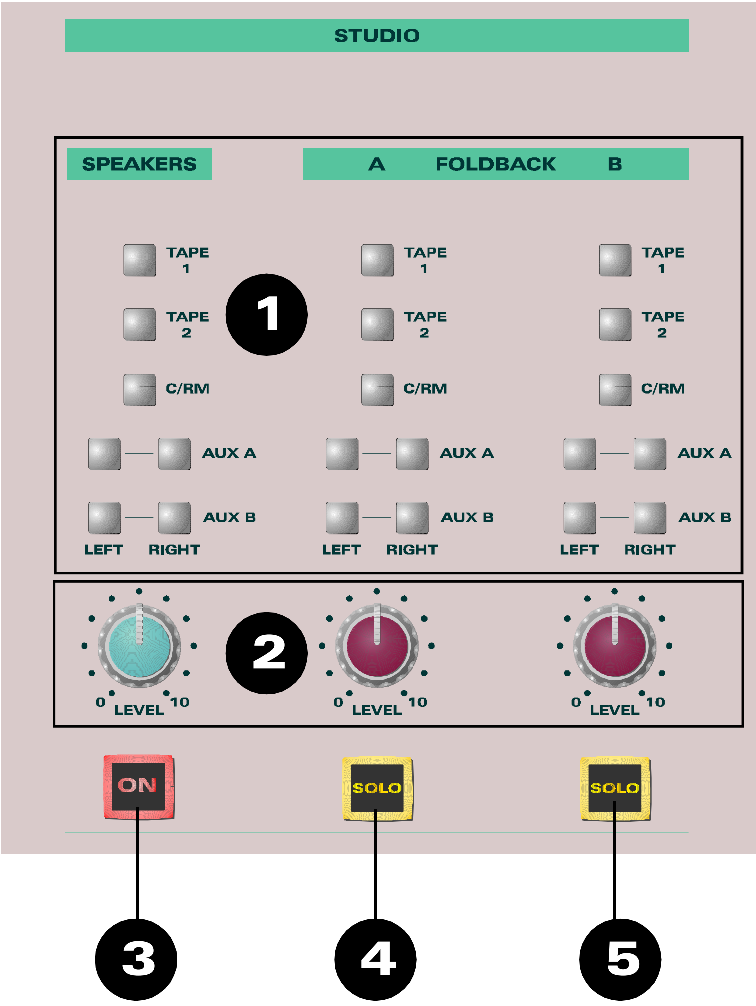
4 **MIC/LINE INSERT RETURN.** When the insert point is in use the signal from the external processing equipment should be connected here.

5 **TAPE INPUT.** The output of the multi-track tape recorder should be connected here although the input is suitable for any high level line source.

6 **TAPE INSERT SEND.** This is the insert point send output for the tape input. Signal is always present here and this can be used as an additional output. Only the Tape signal will appear here and it is not affected by the Flip switch.

7 **TAPE INSERT RETURN.** When the insert point is in use the signal from the external processing equipment should be connected here.

8 **CHANNEL IDENTIFICATION NUMBER.**



STUDIO SPEAKERS and FOLDBACK

This section of the master module looks after the STUDIO LOUDSPEAKER and FOLDBACK outputs of the console. In every case the same sources are available -

TAPE 1

TAPE 2

CONTROL ROOM (C/RM)

AUXILIARY A (AUX A)

AUXILIARY B (AUX B)

Tapes 1 and 2 are external inputs to the console and could be anything you care to plug in. Typically a stereo recorder output is connected here.

C/RM selects the control room monitor output as the source. This is used to provide a playback of the MIX to the studio speakers, or to set up a quick foldback mix, perhaps with a little help from the Aux A and Aux B buses.

AUX A and AUX B which are mono sources which can be independently selected to the left and right sides of the studio speaker or foldback outputs.

Each output has a level control. The studio loudspeaker has a CUT switch and the FOLDBACK outputs have SOLO switches. These enable the foldback mix to be created or checked in the control room. This is always AFL.

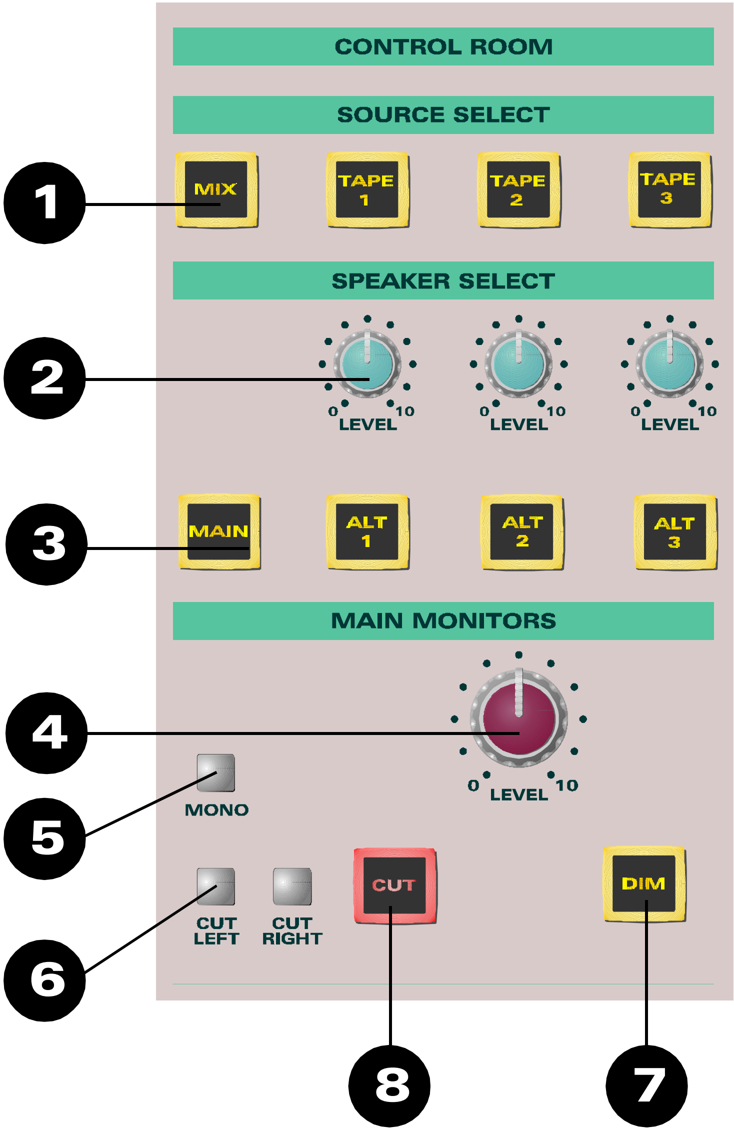
1 SOURCE SELECT. These select the signals to be sent to the studio loudspeakers and foldback.

2 The LEVEL controls set the Studio Loudspeaker and Foldback levels.

3 Studio Loudspeaker ON switch. This mutes the Studio Loudspeaker.

4 FOLDBACK 1 SOLO switch. Allows foldback 1 to be checked on the control room monitoring system.

5 FOLDBACK 2 SOLO switch. Allows foldback 2 to be checked on the control room monitoring system.



1 **SOURCE SELECT** buttons.

2 **LEVEL** controls for the alternate loudspeakers.

3 **SPEAKER SELECT** buttons.

4 Main Loudspeaker **LEVEL** Control.

5 **MONO** button allowing the signal to be checked for mono compatibility.

6 **CUT** buttons allowing independent cutting of the left and right speakers.

7 **DIM** allows the speakers to be reduced in level by a preset amount (15dB).

8 **CUT** allows the speakers to be completely silenced.

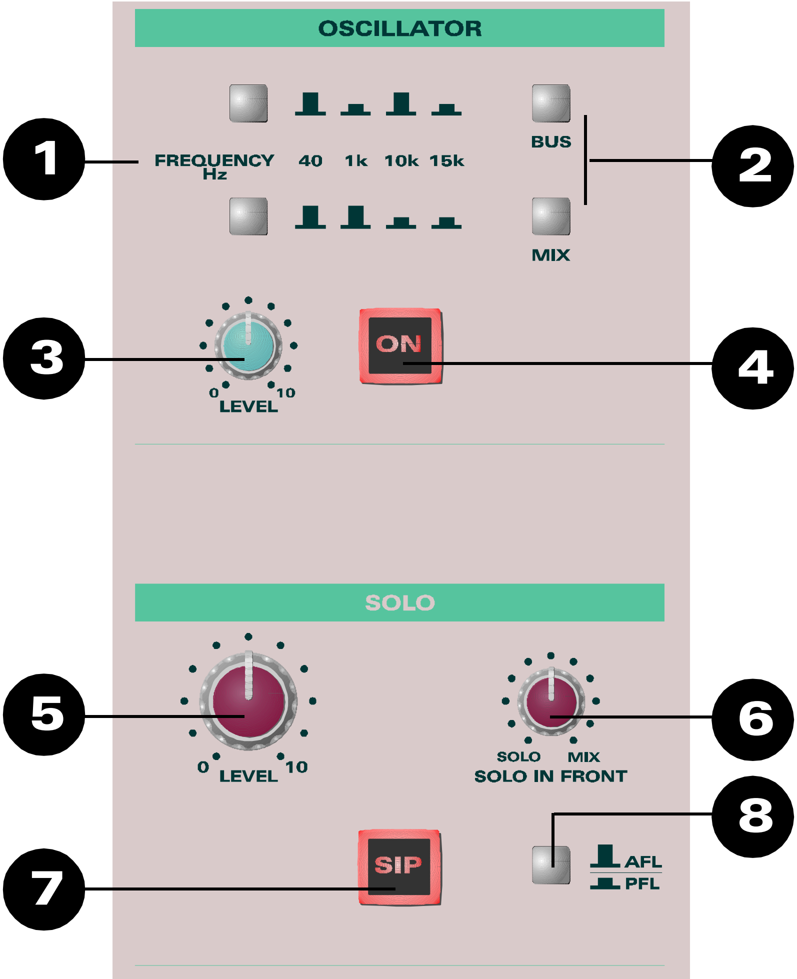
CONTROL ROOM MONITORING

The Control Room Monitoring system is one of the most used systems in a studio. Typically there are several sets of loudspeakers with a main pair and one or more alternate pairs. These are sometimes referred to as Near, Far and Mid field monitors depending on their proximity to the listening position.

The ASP8024 provides for 4 pairs of loudspeakers, a MAIN pair and ALTERNATE 1, 2 and 3. Source selection is made from either the MIX or TAPE 1, 2 and 3.

Each loudspeaker pair has its own level control. The control for the main pair is larger than the controls for the alternate monitors. The level controls are independent of each other.

Associated with the main loudspeakers are some further controls allowing the left and right outputs to be independently or simultaneously CUT, the monitoring system to be placed in MONO mode, or for the output to be dimmed by a preset amount.



1 Oscillator **FREQUENCY** selection switches. Frequencies of 40Hz, 1kHz, 10kHz or 15kHz can be selected using the two switches.

2 Oscillator **ROUTING** switches. These route the oscillator signal directly to either the group buses or the stereo mix bus.

3 Oscillator **LEVEL** Control.

4 Oscillator **ON** switch. During normal operation of the console ensure that the oscillator is switched **OFF**.

5 **SOLO** level control.

6 **SOLO IN FRONT** alters the ratio of the SOLO signal to the MIX signal making it possible to listen to the SOLO'd signal in isolation or combined with some amount of the stereo mix.

7 The **SIP** switch switches the console into SOLO IN PLACE mode. This means that the solo signal will replace the mix signal at the stereo output of the console.

8 This selects either **PFL** or **AFL** mode. This switch is not effective in SOLO IN PLACE mode.

OSCILLATOR

The 4 frequency oscillator can be assigned either to the group buses or to the stereo mix bus. The level is adjustable and when not in use the oscillator should be completely turned off.

SOLO

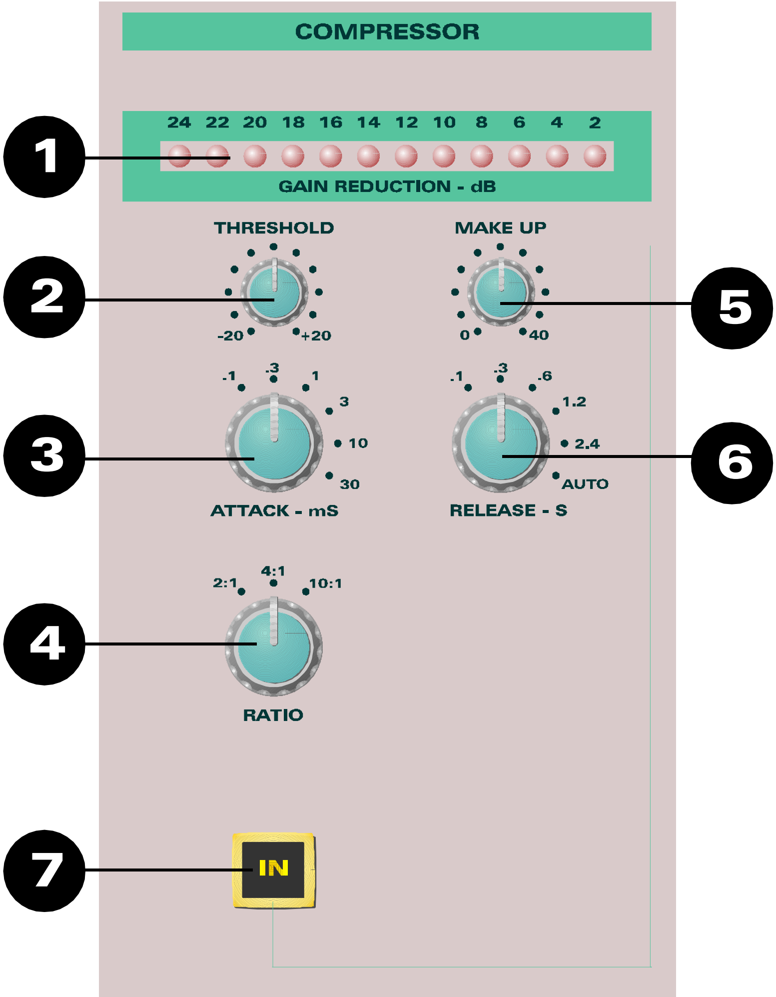
This is the master control area for the AFL/PFL and SOLO IN PLACE system. If SIP is not illuminated then the console will be in either AFL or PFL mode depending upon the PFL/AFL switch.

PFL allows pre fade signals to be auditioned on the monitors and viewed on the master meters by pressing the SOLO button on the appropriate channel strip.

AFL allows the post fade post pan signals to be auditioned on the monitors and viewed on the master meters by pressing the SOLO button on the appropriate channel strip.

A control adjusts the relative level of the solo signal and the mix signal allowing the solo'd channel to be heard in isolation or with some amount of the mix.

SIP mode allows the post pan signal to be auditioned by switching the solo'd channels onto the console output and muting the stereo mix for the duration of the solo. Since this happens on the mix output, solo in place should not be used during mixing but is very helpful when setting up the mix.



1 A 12 segment bargraph **METER** indicating the amount of gain reduction taking place.

2 **THRESHOLD** control for determining the level where compression starts. Threshold can be set between -20dBu and +20dBu.

3 **ATTACK** controls the speed with which the compressor reacts to the signal. Attack is adjustable in 6 steps from 0.1 to 30ms.

4 **RATIO** determines the amount of compression used once the signal is above the threshold level. High ratios means that more compression will be applied. For example a ratio of 2:1 means that the output will rise 1dB for every 2dB rise in input level. Ratios of 4:1 and 10:1 are also available.

5 **MAKEUP** is a gain control. When a signal is compressed the level is reduced. A gain make up stage restores the peak level although of course the dynamic range of the signal has now been reduced. 40dB of gain make up is available.

6 **RELEASE** controls the speed at which the compressor allows the gain to return to normal when the signal drops below the threshold level. Release is available from 0.1s to 2.4s in 5 steps with an additional AUTO setting.

7 The **IN** switch is used to switch the compressor into circuit.

COMPRESSOR

A compressor can be switched into the main stereo signal path when required. Note that it is located after the mix insert point but before the main fader.

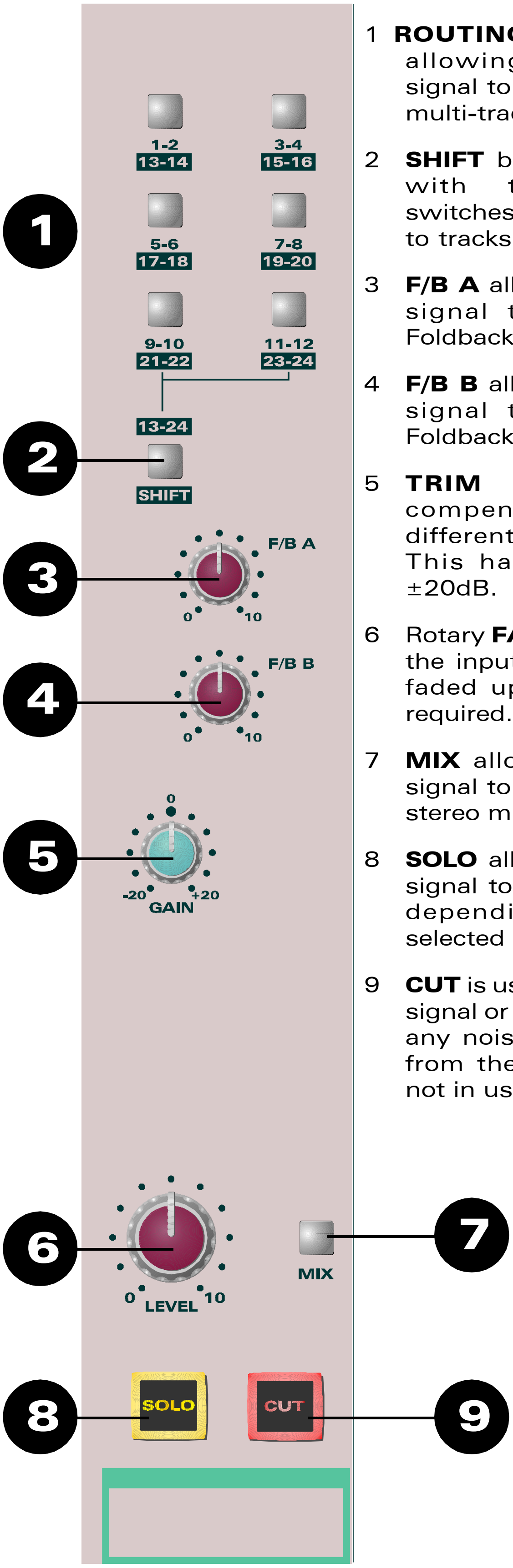
The compressor characteristics are optimized for use in mix processing while many of the parameters remain under the control of the engineer.

Threshold, Gain Make Up, Attack, Release and Ratio are all adjustable while the entire processor can be switched out of circuit when not required.

A bargraph meter indicates the amount of compression applied to the signal. When compression is taking place this should show some very healthy activity. Please do not try to over compress the mix output as the results will sound terrible. Always use your ears to check the effect that you are trying for!

You too can act as a compressor by reducing and increasing the master fader level. In some situations this gain riding may be preferable as, assuming you know the song, you will be able to anticipate peaks and reduce the gain slowly before they occur, increasing it slowly as the signal level falls from the peak.

A good use of the compressor may be to limit peak signals by setting a highish threshold with a high ratio. Thus when the threshold is reached the signal is barely allowed to increase beyond it.



1 **ROUTING SWITCHES** allowing the input signal to be sent to the multi-track recorder.

2 **SHIFT** button working with the routing switches to give access to tracks 13 - 24.

3 **F/B A** allows the input signal to be fed to Foldback A.

4 **F/B B** allows the input signal to be fed to Foldback B.

5 **TRIM** control to compensate for different input levels. This has a range of ±20dB.

6 Rotary **FADER** allowing the input signal to be faded up or down as required.

7 **MIX** allows the input signal to be sent to the stereo mix bus.

8 **SOLO** allows the input signal to be auditioned depending upon the selected solo mode.

9 **CUT** is used to mute the signal or simply remove any noise contribution from the return when not in use.

STEREO INPUTS

The Stereo inputs allow signals to be brought back into the console (returned) from an effect unit without using up a complete channel strip. Often stereo signals can only be routed to the stereo mix, however, the ASP8024 stereo inputs allow routing back to the multi-track in addition to the mix. This means that effects can be recorded on the multi-track.

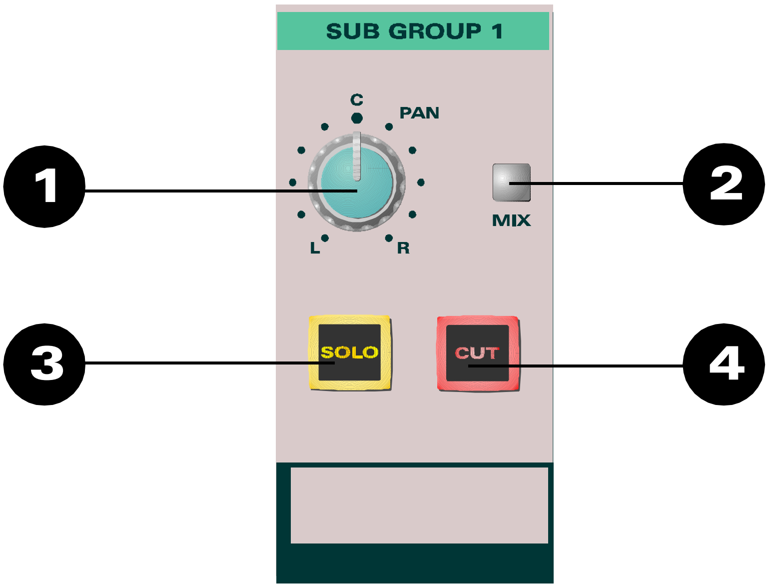
Routing to the foldback system is also possible using F/B A and F/B B allowing performers to hear any reverberation or other effect.

A gain trim with 20 dB of range, rotary fader, solo and cut switch complete the facilities on the input.

The stereo channels effectively increase the number of inputs on the console that are available especially during a mix down. Every channel has two inputs, either mic/line or tape, thus a 36 input console has 72 inputs PLUS 4 stereo inputs giving 80 inputs.

SUB GROUP OUTPUTS

In addition to routing signal to the group outputs for recording on a multi-track recorder, the first eight groups are also fed to 8 sub group outputs. Each sub group has a PAN control, SOLO and CUT switches and a fader controlling the group output level. The sub groups have insert points located on the rear panel of the console allowing external processing to be patched in when called for. The MIX switch assigns the sub group to the stereo mix bus and allows the sub groups to be used in mix down. The subgroup INSERT RETURNS can be used as extra inputs to the mix.



1 The **PAN** knob pans the subgroup signal across the stereo bus.

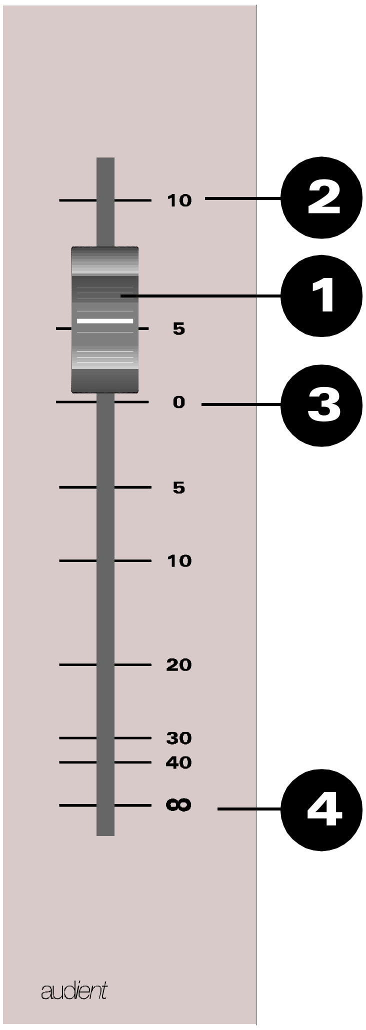
2 **MIX** allows the sub group to be assigned to the stereo mix bus and hence the main console output.

3 **SOLO** allows the subgroup signal to be auditioned depending upon the selected pan mode.

4 **CUT** mutes the subgroup output.

SUB GROUP FADER

The 8 subgroup faders are located in the centre of the console, close to the operator, and control the subgroup output levels. These outputs are directly available at the rear of the console or they may be used to feed the stereo mix bus simply by pressing the mix buttons.

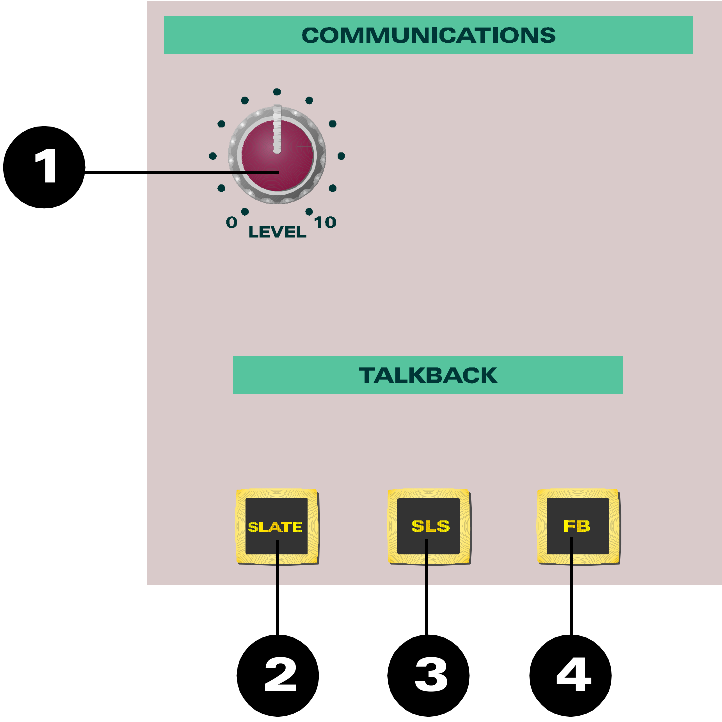


1 The **SUB GROUP FADER** sets the level of the subgroup output signal.

2 This is the **+10db** mark. The fader is fully open at this point and introducing 10dB of gain into the signal path.

3 This is the **0dB** mark which is the normal operation position for the fader.

4 With the fader here the signal path is closed and no signal will pass.



1 The **LEVEL** control adjusts the talkback level.

2 Pressing **SLATE** allows talkback to the group outputs. This allows track identification information, for example, to be recorded.

3 Pressing **SLS** lets you talk to the studio loudspeaker.

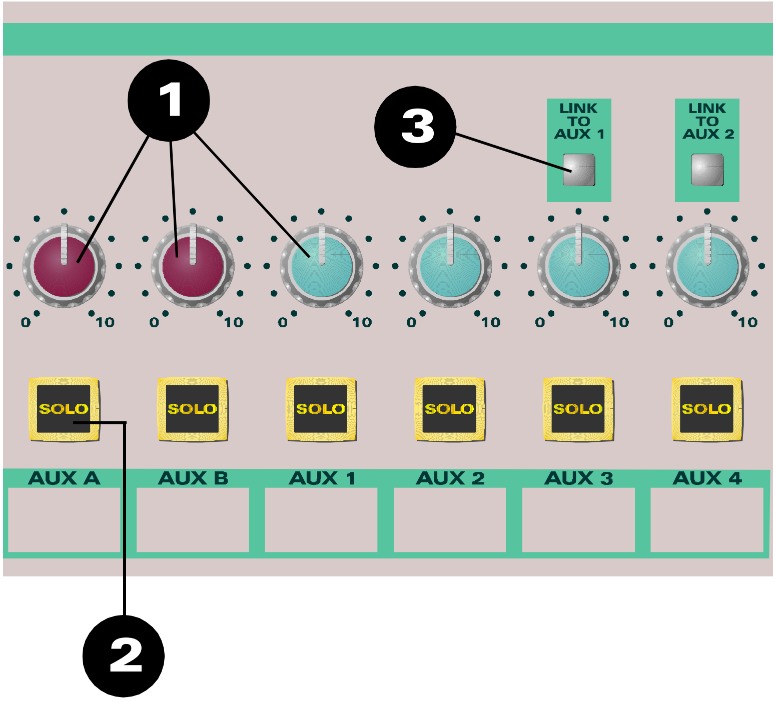
4 Pressing **FB** lets you talk to the Foldback outputs.

In all cases pressing a talkback key will cause the control room monitors to dim. This helps prevent feedback, makes the talkback more intelligible and does not affect the console outputs.

TALKBACK

Talkback is used to communicate with the STUDIO, the FOLDBACK system or the GROUP outputs of the console. Note that the talkback to the foldback system will work even when the foldback levels are turned down.

The talkback microphone may be phantom powered by connecting Link 1 on the PC10801 board.



1 **AUXILIARY MASTER LEVEL** controls the overall auxiliary output.

2 **SOLO** button used to audition the auxiliary output. This is always AFL.

3 The **LINK** button is used when it is required to combine signals from different auxiliaries into a common signal. The leftmost auxiliary becomes the overall master, thus if auxiliary 3 is linked to auxiliary 1 then the output of auxiliary 1 should be used as the master output.

AUXILIARY MASTERS

The auxiliary masters control the overall level of the auxiliary outputs. A balance or mix can thus be created by using the controls on the channel strips and the overall level adjusted by using the auxiliary master control. The auxiliary outputs can be solo'd so that a balance can be created by listening to the output.

A typical mixing situation may require that the LF and SF paths feed the same effect device.

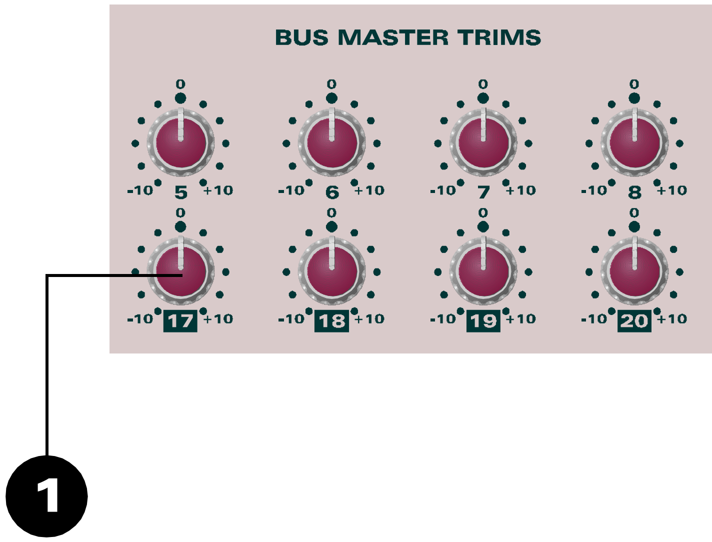
Although an auxiliary send can be assigned to either of the two paths they naturally cannot access both paths at the same time. By linking auxiliaries this problem is overcome and if the auxiliary 3 signals need to be combined with those on Auxiliary 1 this can be achieved by simply using the link facility. The following combinations are possible:

AUXs 3 and 5 can be linked to AUX 1.

AUX s 4 and 6 can be linked to AUX 2.

AUXs 9 and 11 can be linked to AUX 7.

AUXs 10 and 11can be linked to AUX 8.

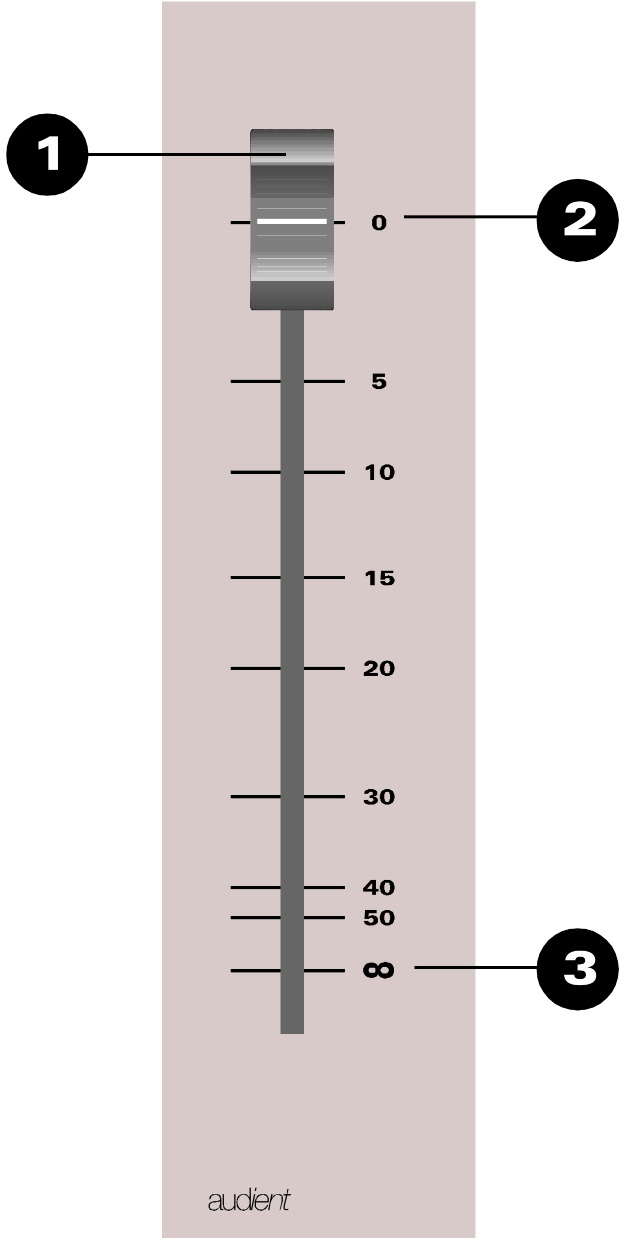


BUS MASTER TRIM

The BUS MASER TRIMS are the final stage of level control over the signals routed to the group outputs. Each group has a bus trim although for the purposes of this manual only a few are shown in the accompanying diagram.

The trim range is ±10dB.

1 **BUS MASTER TRIM** controls the overall level of a group output.



1 The **MASTER FADER** controls the level of the stereo output signal.

2 This is the **0db** mark. The fader is fully open at this point.

3 With the fader here the signal path is closed and no signal will pass.

MASTER FADER

The master fader is used to control the stereo output of the console. Unlike the channel faders it is calibrated with the 0dB mark at the top as the main purpose of this fader is to create a fade out at the end of a title.

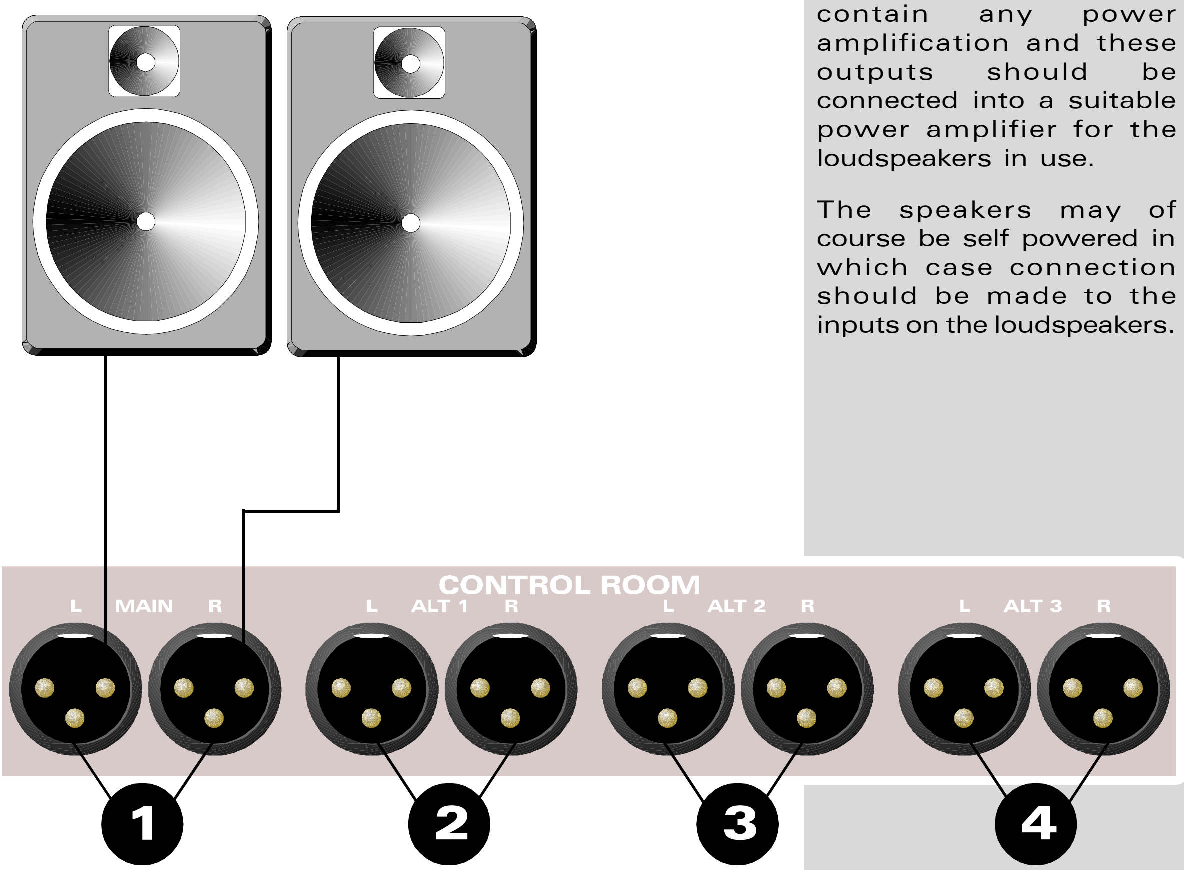
Under normal operating conditions the fader should always be set at maximum. If it has to be pulled back a significant distance it indicates that the levels to the mix bus are too high and should be reduced.

CONTROL ROOM CONNECTORS

These are the connectors for the control room loudspeakers. There is a main output and 3 alternative outputs which can be selected from the control surface of the console. Each output is stereo, having a left and right connector.

The console does not contain any power amplification and these outputs should be connected into a suitable power amplifier for the loudspeakers in use.

The speakers may of course be self powered in which case connection should be made to the inputs on the loudspeakers.



1 **MAIN** Control room loudspeaker output.

2 **ALTERNATIVE 1** loudspeaker Output.

3 **ALTERNATIVE 2** loudspeaker Output.

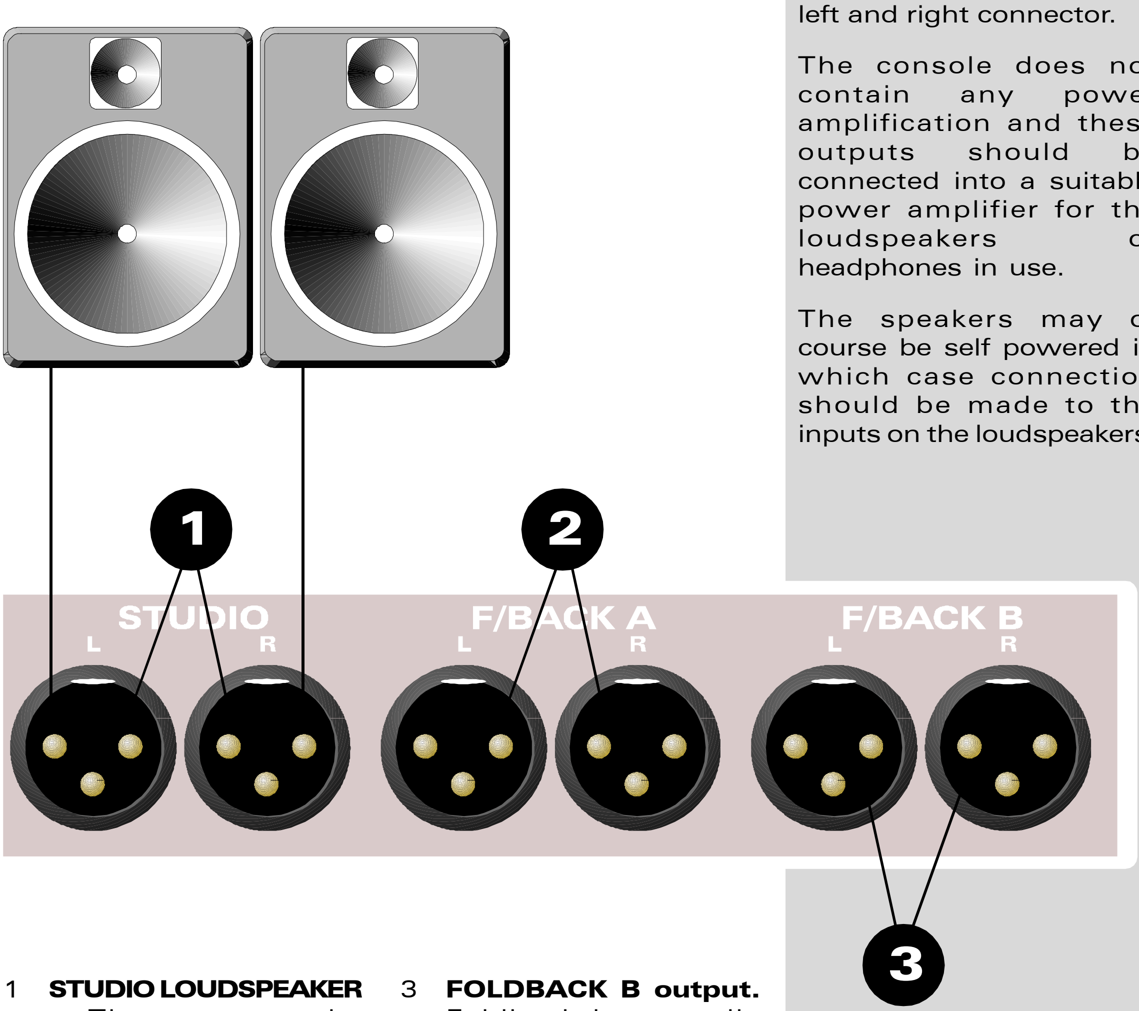
4 **ALTERNATIVE 3** loudspeaker Output.

STUDIO AND FOLDBACK CONNECTORS

These are the XLR connectors for the Studio Loudspeaker and the two foldback outputs. Each output is stereo, having a left and right connector.

The console does not contain any power amplification and these outputs should be connected into a suitable power amplifier for the loudspeakers or headphones in use.

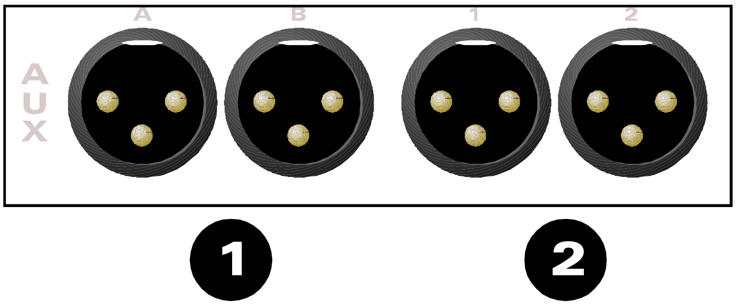
The speakers may of course be self powered in which case connection should be made to the inputs on the loudspeakers.



1 **STUDIO LOUDSPEAKER**. These carry the outputs to the studio loudspeaker system. The output depends upon the selection made on the control surface of the console.

2 **FOLDBACK A output.** Foldback is normally used as a feed to headphones. The output depends upon the selection made on the control surface of the console.

3 **FOLDBACK B output.** Foldback is normally used as a feed to headphones. The output depends upon the selection made on the control surface of the console.



1 **AUXILIARY A and AUXILIARY B outputs.** These auxiliaries are normally used as feeds to the foldback system by selecting them on the control surface of the console. They can also be used as additional effects sends.

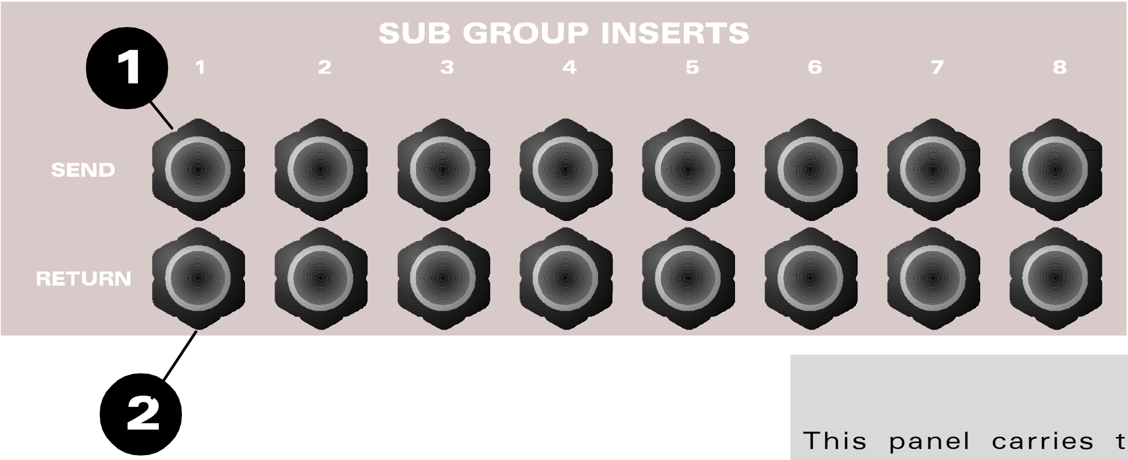
2 **AUXILIARY 1 and 2 outputs.** The remaining auxiliary outputs are identical.

AUXILIARIES CONNECTORS

Not all of the ouputs are shown for clarity. It carries XLR connectors for Auxiliary A, Auxiliary B and Auxiliaries 1 through to 12. (Note that all auxiliaries are mono.)

The auxiliary outputs are typically used to send to effect units such as reverberation units. They are line level fully balanced outputs.

SUB GROUP INSERT CONNECTORS



This panel carries the connectors for the sub group insert points. There are 8 sub groups and each insert point has a send output and a return input.

Signal is always present on the send output. If required the insert returns could be used as very basic inputs to the stereo mix bus during mixdown, from a submixer or sampler etc..

1 **INSERT SEND** Jack for sub group 1

2 **INSERT RETURN** Jack for sub group 1

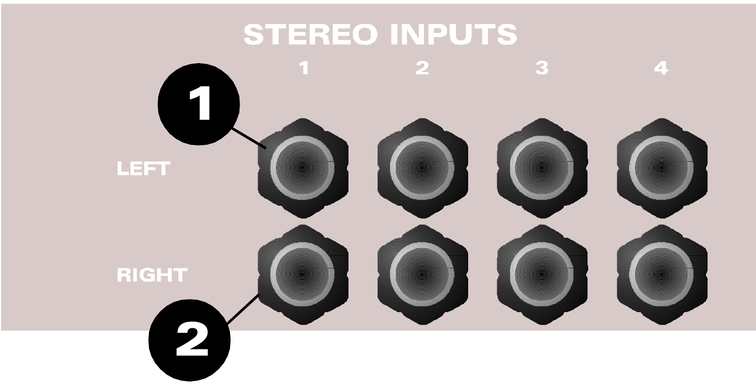
The 7 remaining subgroups have identical connectors.

STEREO INPUT CONNECTORS

This panel carries the input TRS jacks for the stereo inputs located on the master section of the console.

There are 4 returns each with a left and a right input.

If a mono source is used then it should be plugged into the left input. This is normalled over to the right input causing the signal to travel through the left and right signal paths of the return. If a plug is inserted into the right input the normalled connection is broken.



1 **LEFT INPUT** for STEREO INPUT 1.

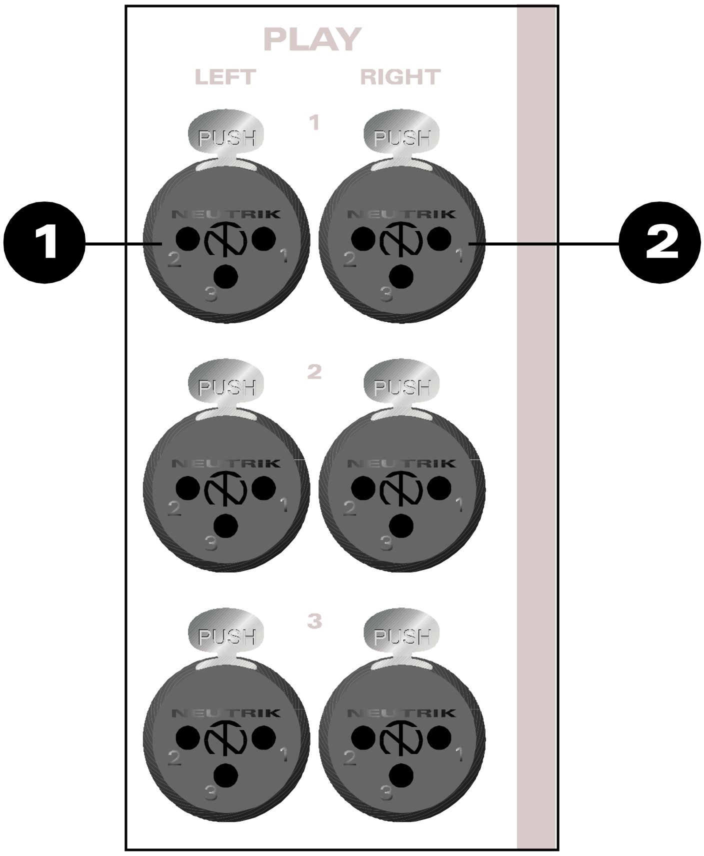
2 **RIGHT INPUT** for STEREO INPUT 1.

The remaining 3 Returns have identical connectors.

PLAY INPUTS

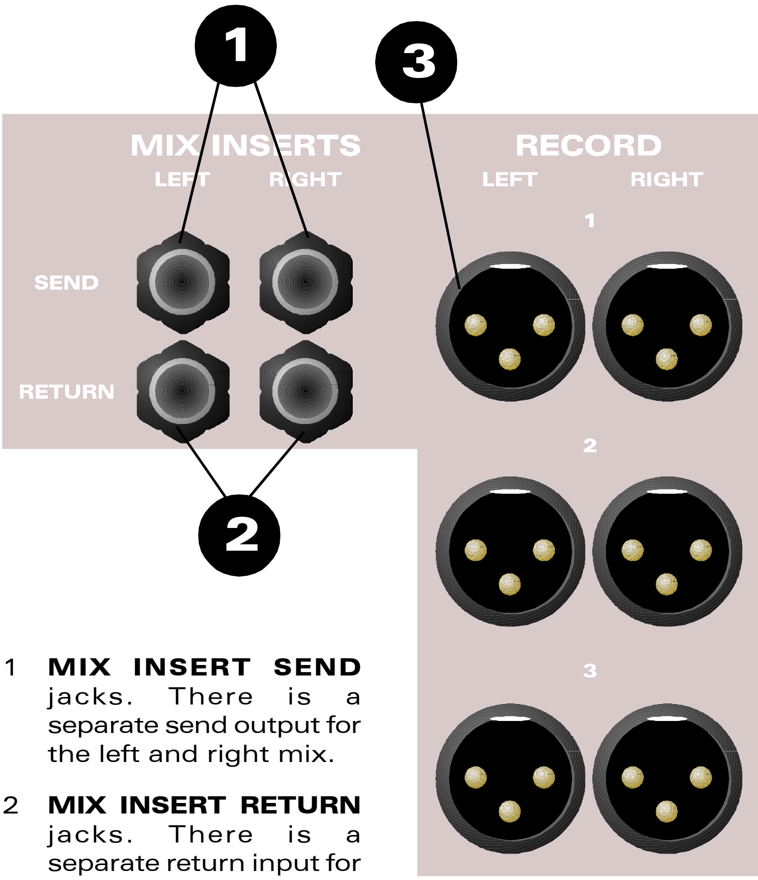
This panel carries the Play Inputs which are external inputs to the monitoring system.

All three inputs are available as sources to the control room monitoring system while the first two can be used as sources for the studio monitoring system and the foldback system.



1 **Left Input** for Play 1.

2 **Right Input** for Play 1.



1 **MIX INSERT SEND** jacks. There is a separate send output for the left and right mix.

2 **MIX INSERT RETURN** jacks. There is a separate return input for the left and right mix.

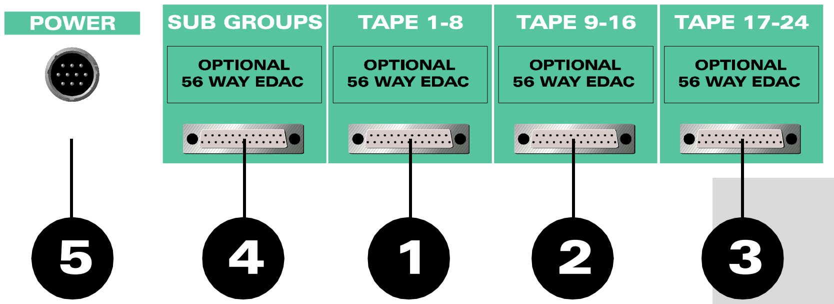
3 **MIX OUTPUT** (Stereo Output) connectors. There are three sets of connectors allowing the connection up to 3 stereo recorders.

MAIN OUTPUTS

There are 3 sets of stereo output connectors enabling the connection of 3 stereo recording machines. The outputs can be used for many purposes of course and it is not essential that they feed a recorder.

Associated with the outputs are the mix insert points. They allow a final mix processor to be inserted into the MIX path, and are located before the MIX compressor/limiter.

MULTI-TRACK CONNECTORS



1 **MULTI-TRACK CONNECTOR** for tracks 1 through 8.

2 **MULTI-TRACK CONNECTOR** for tracks 9 through 16.

3 **MULTI-TRACK CONNECTOR** for tracks 17 through 24.

4 **SUB-GROUP OUTPUT CONNECTOR.**

5 **POWER INPUT CONNECTOR.**

This panel contains the connectors for the multi-track tape recorder. Rather than having individual connectors for each group output it is much more convenient (and quicker to make a connection) if they are on multi-pole connectors.

The multi-track recorder sends are split across 3 connectors with each carrying the group outputs for 8 tracks. The sub-group outputs are on a fourth connector.

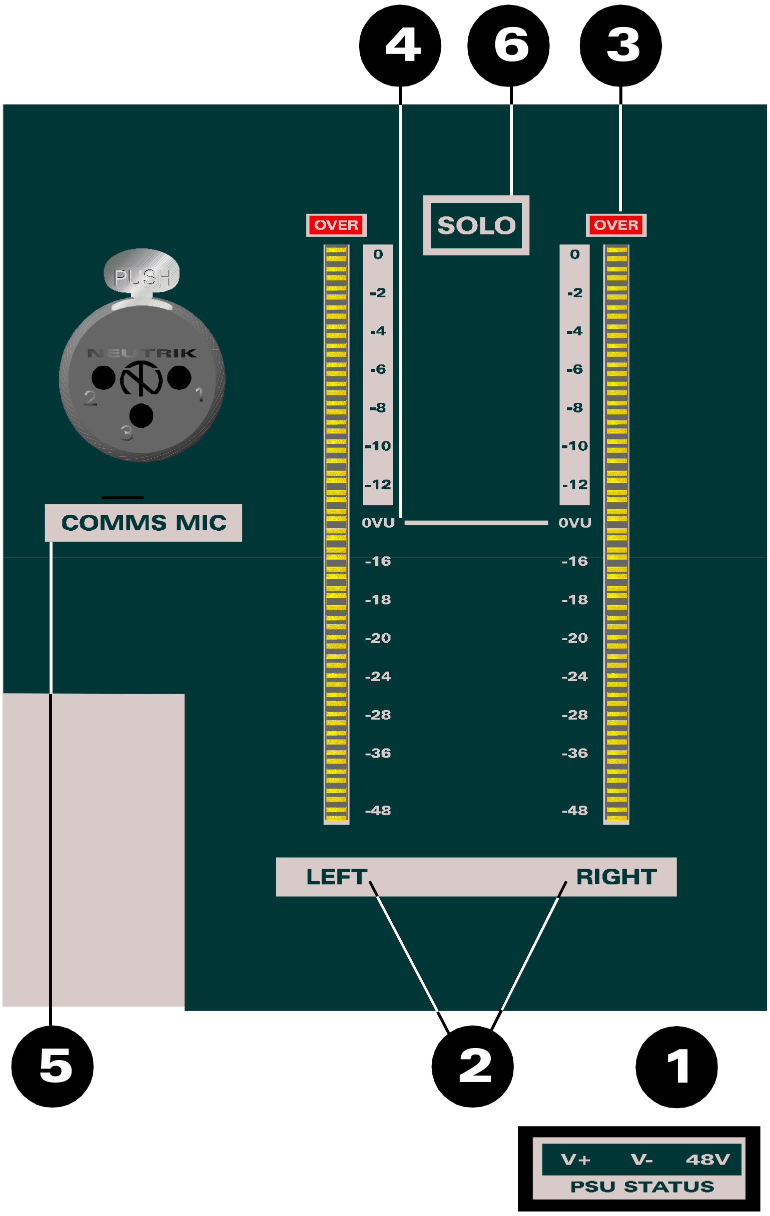
The multitrack sends and sub group outputs are also available (as a cost option) on 56 pin EDAC multi-pole connectors.

MULTI-PIN CONNECTIONS						
SIGNAL NUMBER	+VE SIGNAL		-VE SIGNAL		SCREEN	
	D-SUB PIN	EDAC PIN	D-SUB PIN	EDAC PIN	D-SUB PIN	EDAC PIN
1/9/17	24	A	12	E	25	F
2/10/18	10	C	23	B	11	H
3/11/19	21	D	9	K	22	J
4/12/20	7	P	20	V	8	N
5/13/21	18	Z	6	d	19	U
6/14/22	4	c	17	f	5	Y
7/15/23	15	j	3	n	16	m
8/16/24	1	t	14	y	2	s

Note: All undesignated pins are unconnected. All screen connections are joined inside the console and connected to metalwork earth.

Patchbays: Tie lines connections 25-32 etc follow the same wiring convention shown above. Consoles fitted with patchbays do not have the option of EDAC connectors.

THE MASTER METERS



Metering has been addressed on the input modules although there are also meters associated with the master functions on the console.

The output meters show the output levels of the subgroups and also the stereo output. If problems are indicated here with the level being either too high or low then it is most likely that the problem is further back in the signal path. Follow our procedure for gain setting to see if this remedies the situation.

1 **Power Supply Status** Indicators for the +18V, -18V and 48V power rails.

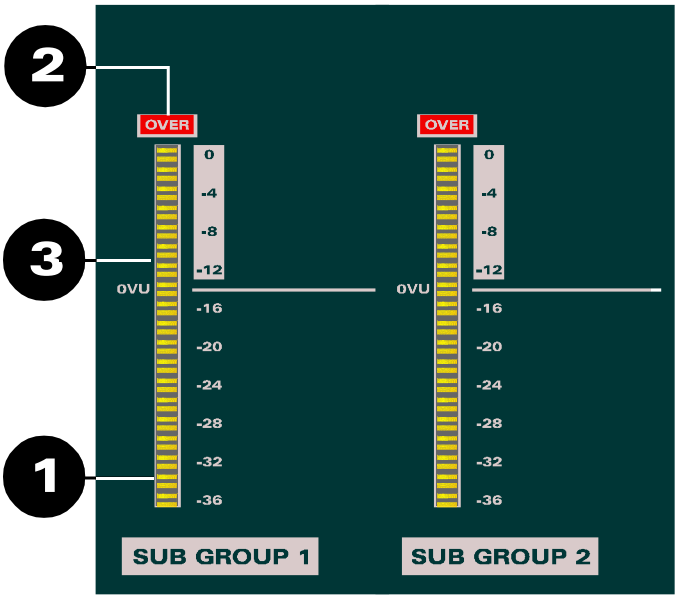
2 Left and Right Peak Reading **Main Output Meters.**

3 **Over Indicators,** when illuminated the signal is in danger of being clipped and the level should be reduced. A level of +20dBu will bring the **OVER** indicator on. Maximum or 0dBFS is indicated at a signal level of +18dBu, corresponding to full modulation on a digital recorder.

4 **0VU.** When the signal is at a level of +4dBu the bars should be illuminated up to this point. Note that this is not meant to be normal operating level because the meters are peak reading.

5 **Talkback Microphone Connector.** If talkback is to be used then a microphone should be plugged in here.

6 **Solo Indicator.** This lights when a SOLO button has been pressed on the console.



1 **Sub Group Meter.** There are 8 peak reading subgroup meters - 1 for each sub group output.

2 **Over Indicator.** When illuminated the signal is in danger of being clipped and the level should be reduced. Like the main meters this will illuminate with a signal level of +20dBu. 0dBFS is indicated at a level of +18dBu, which normally corresponds to full modulation on a digital recorder.

3 **0VU.** When the signal is at a level of +4dBu the bars should be illuminated up to this point. Note that this is not meant to be normal operating level because the meters are peak reading.

THE SUB GROUPS

There 8 sub group outputs on the console (only 2 have been shown for clarity). The sub groups are accessed in parallel with the main group outputs and only the first 8 groups therefore have a corresponding sub group.

The sub groups are able to feed the stereo mix and also have their own outputs located on the rear panel of the console.

The sub group outputs can be directly used to feed an 8 track recorder.

A typical use for the sub groups would be during mixdown when a number of signals are to be combined to one fader for simpler control. If the signals are on the LF path then they must first of all be sent to the routing matrix by pressing the button in the routing section. By routing the signals to groups 1 through 8 they will also be routed to the sub groups.

THE PATCHBAY

The ASP8024PB optional patchbay system provides front panel access to the connections provided on the rear of the standard console as well as a number of additional ‘patch points’.

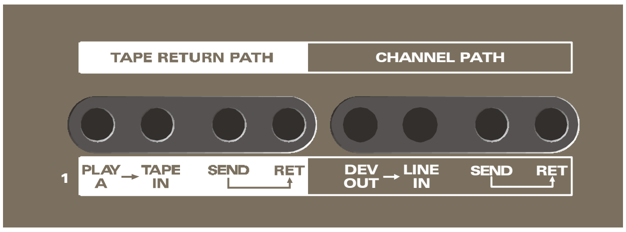
Patching is implemented using TT size jacks with D-sub connectors provided for the studio systems interfaces.

All default signal paths are fully normalled to avoid unnecessary patching and functional patch point blocks are arranged in a horizontal format for ease of identification.

Interfacing for two 24track machines is provided and all patchbays are supplied with 144 tie-lines as standard.

1. TAPE RETURN PATH

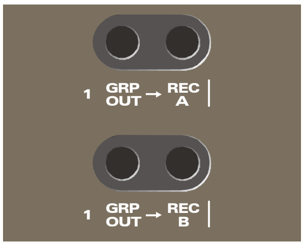
2. CHANNEL PATH



Provision is made for interfacing to two 24-track machines. Tape machine A playback is normalled from the relevant D-sub interface to the Tape inputs of channels 1-24. Tape machine B playback is normalled to channels 25 onwards up to 48 if this many channels are fitted.

A rear panel jack socket is provided for interfacing external devices such as synthesizers. This signal is available as the Device Output on the patchbay and is normalled to the channel line input.

3. MULTITRACK SENDS

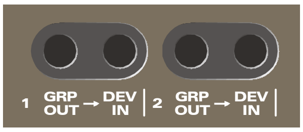


The sends to both 24 track machines are normalled from the 24 bus outputs. The machine sends may of course be overpatched from channel insert sends, sub-groups etc.

4. STEREO INPUTS

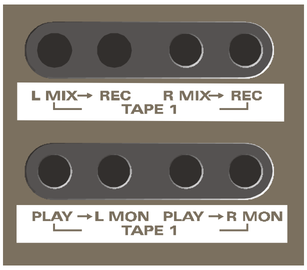
The stereo device outputs such as reverbs etc are interfaced via 1/4 inch jack sockets on the rear of the console. These are normalled to the 4 stereo inputs through the patch jacks shown above.

5. SUB GROUPS



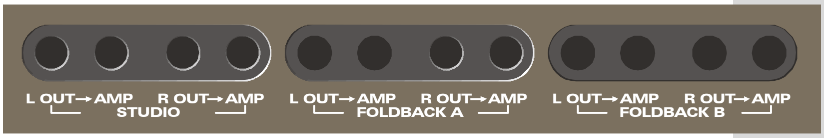
The eight Sub Group outputs are normalled to a D-sub connector and may be used for providing outputs to a device such as a non-linear editing system for example.

6. STEREO TAPES



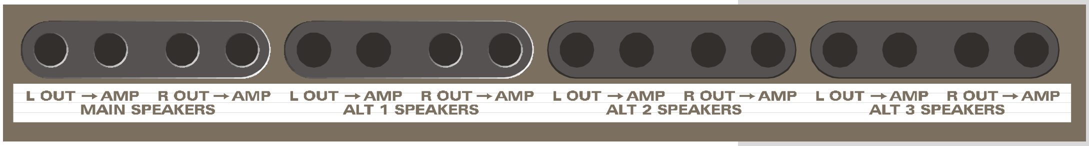
The stereo tape machines are interfaced on XLR connectors fitted to the rear of the console. The feeds to and from the XLRs are normalled via the patch bay from the Mix output and to the 2 track monitor returns respectively.

7. **STUDIO/FOLDBACK**



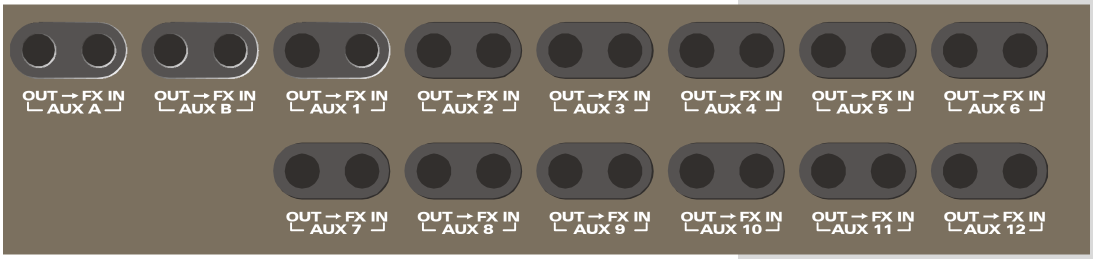
The outputs to the foldback and studio speaker amplifiers are terminated on XLRs on the rear of the console and are normalled from the relevant console sends via the patchbay.

8. **CONTROL ROOM**



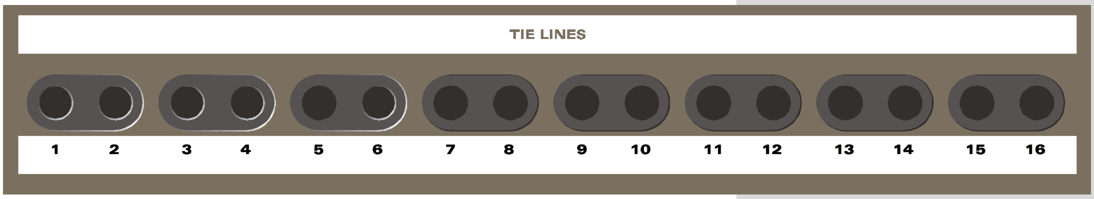
Feeds to the control monitor speaker amplifiers are normalled via the patchbay to XLRs on the rear of the console.

9. **AUXILIARIES**



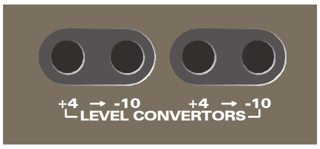
Feeds to external processing devices are normalled from the console aux outputs to rear mounted XLRs

10. **TIE LINES**



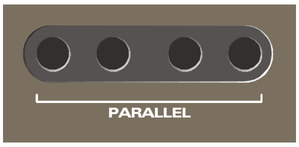
Tie lines are terminated on D-sub connectors on the rear of the console. Pairs of tie lines (1-2) etc may be normalled by fitting two wire links in the positions behind each pair of jacks on the tie line PCBs.

11. **LEVEL CONVERTORS**



Two pairs of passive level convertors are provided. A low impedance output signal plugged into the +4dBu jack will be attenuated to -10dBV suitable for driving the inputs of semi-professional equipment.

12. **PARALLELS**



Two sets of 4 way parallels are provided. These are useful for paralleling a number of inputs. Do not use the parallels to join outputs as this will short the outputs together.

PATCHBAY CONNECTOR PANELS

Connections to studio systems on consoles fitted with the ASP8024PB patchbay system are made via the rear and bottom connector panels generally in a similar fashion to standard consoles. The exceptions to this are dealt with here.

1. **CHANNEL CONNECTIONS**

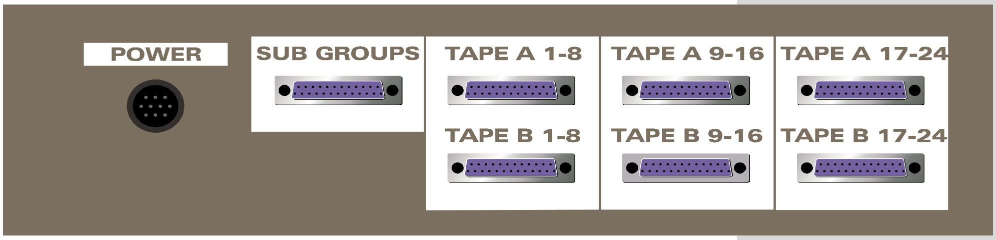
   Microphone input (XLR) and Line input (jack) connectors are located on the rear connectors panel behind each of the input strips as on the standard console (see page 28). The Tape input and insert send/return jacks are omitted.

2. **MASTER CONNECTIONS**

   XLR and Jack connectors associated with the master module are arranged in the same way as the standard console (see pages 39-45) with the exception of the sub-group insert send/return jacks which are omitted as these are included on the patchbay.

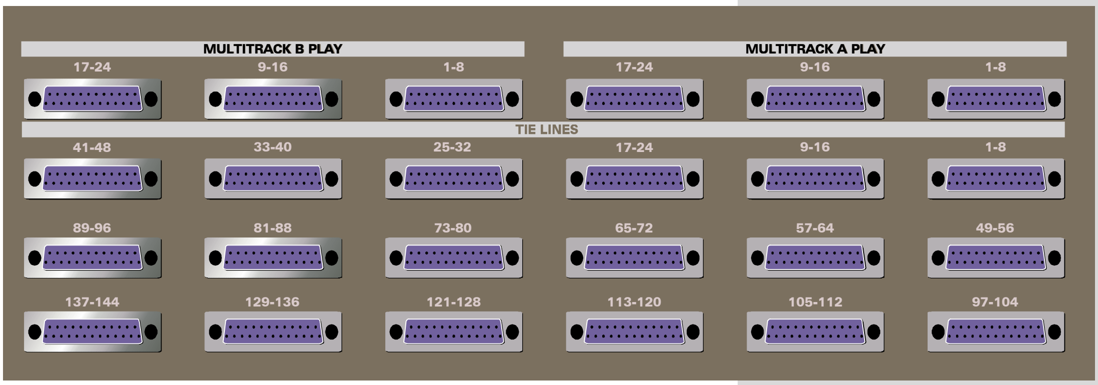
3. **MULTITRACK RECORD/ SUB GROUP CONNECTIONS**

   These are located on the underside of the console adjacent to the PSU connector. All console interfaces are D-sub female types wired to the Tascam DA88 convention with 8 signals per connector (see page 46).



4. **MULTITRACK PLAY AND TIE LINE CONNECTIONS**

   These are located on the rear panel behind the patchbay section. All connectors are again D-sub females wired to the Tascam DA88 convention.



THE POWER SUPPLY

The ASP8120 power supply is a 3U Rack Mounting, unit. There are no controls other than the ON/OFF switch, a reset switch for the +/- 18v rails and the voltage selector located on the rear panel. 3 LEDs are used to indicate the existence of the +18V, -18V and +48V power rails. There are also PSU status indicators located on the master meter section of the console.

Two very low noise slow running fans cool the unit so that it can normally be located in the same room as the console.

A 4 metre long power supply cable is supplied connecting the console to the power supply.

IMPORTANT!

It is important that the power supply ventilation slots and fans are not obstructed. There is no need to allow space above and below the unit if it is mounted in a rack and it may also be placed directly on a floor if required.

Before using please check that the voltage selector on the back panel of the PSU is set correctly for the local mains supply.

SPECIFICATIONS

FREQUENCY RESPONSE

Mic input to Mix output	<+0,-0.3dB	20Hz-20kHz	@6-40dB gain.
Line input to Mix output	<+0,-0.3dB	20Hz-20kHz	@0dB gain.

THD AND NOISE AT +20dB OUTPUT

Mic XLR input to any output	<0.005% at 1kHz
Line input to any output	<0.005% at 1kHz
Tape input to any output	< 0.003% at 1kHz

NOISE

Mic EIN (20-20kHz, 150R source)	<-127.5dBu
Bus noise (no inputs routed)	<-93dBu
Bus noise (36 inputs routed)	<-78dBu

CROSSTALK AND MUTE ATTENUATION AT 1kHz

Short fader Mute	>90dB
Long fader Mute	>90dB
Mix assign	>90dB
Bus assign	>90dB

MIC CMRR

	70dB (Min gain)
	75dB (Max gain)

MAXIMUM INPUT

Mic	>+21dBu (min gain)
Line	>+30dBu (min gain)
insert return	>+21dBu

MAXIMUM OUTPUT INTO 2K OHMS

Mix output	>+26dBu
Bus output	>+26dBu
Aux output	>+26dBu
Insert send	>+20dBu
Monitor, Studio, F/B outputs	>+20dBu

BLOCK DIAGRAMS

1.INPUT STRIP

2.PATCHBAY VERSION INPUT STRIP

3.MASTERS PART 1

4.PATCHBAY VERSION MASTERS PART 1

Stereo and Solo Buses

2 track returns

Solo system

Control room monitoring

5.MASTERS PART 2

6.PATCHBAY VERSION MASTER PART 2

Stereo inputs

Aux Buses

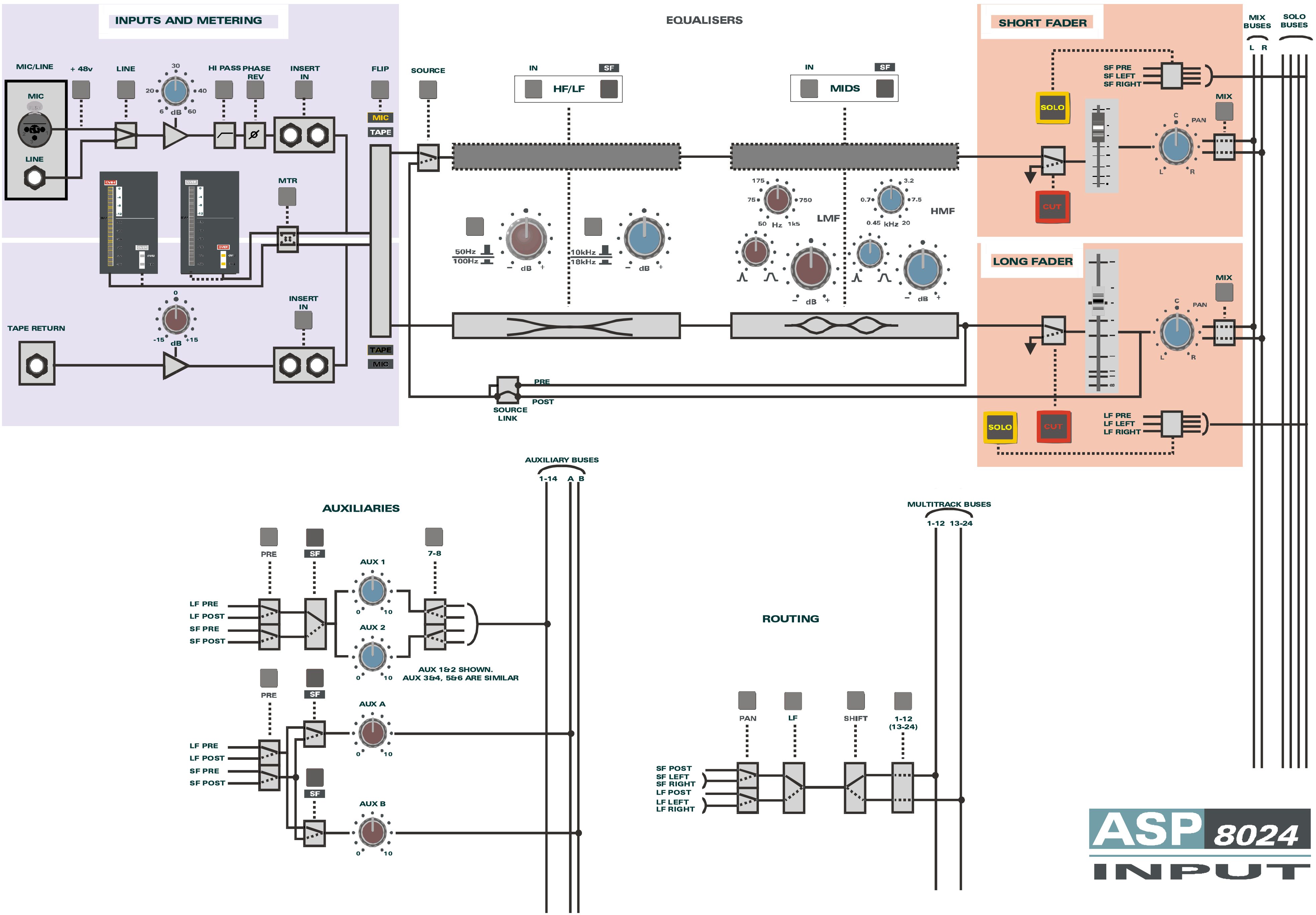
Multitrack Buses

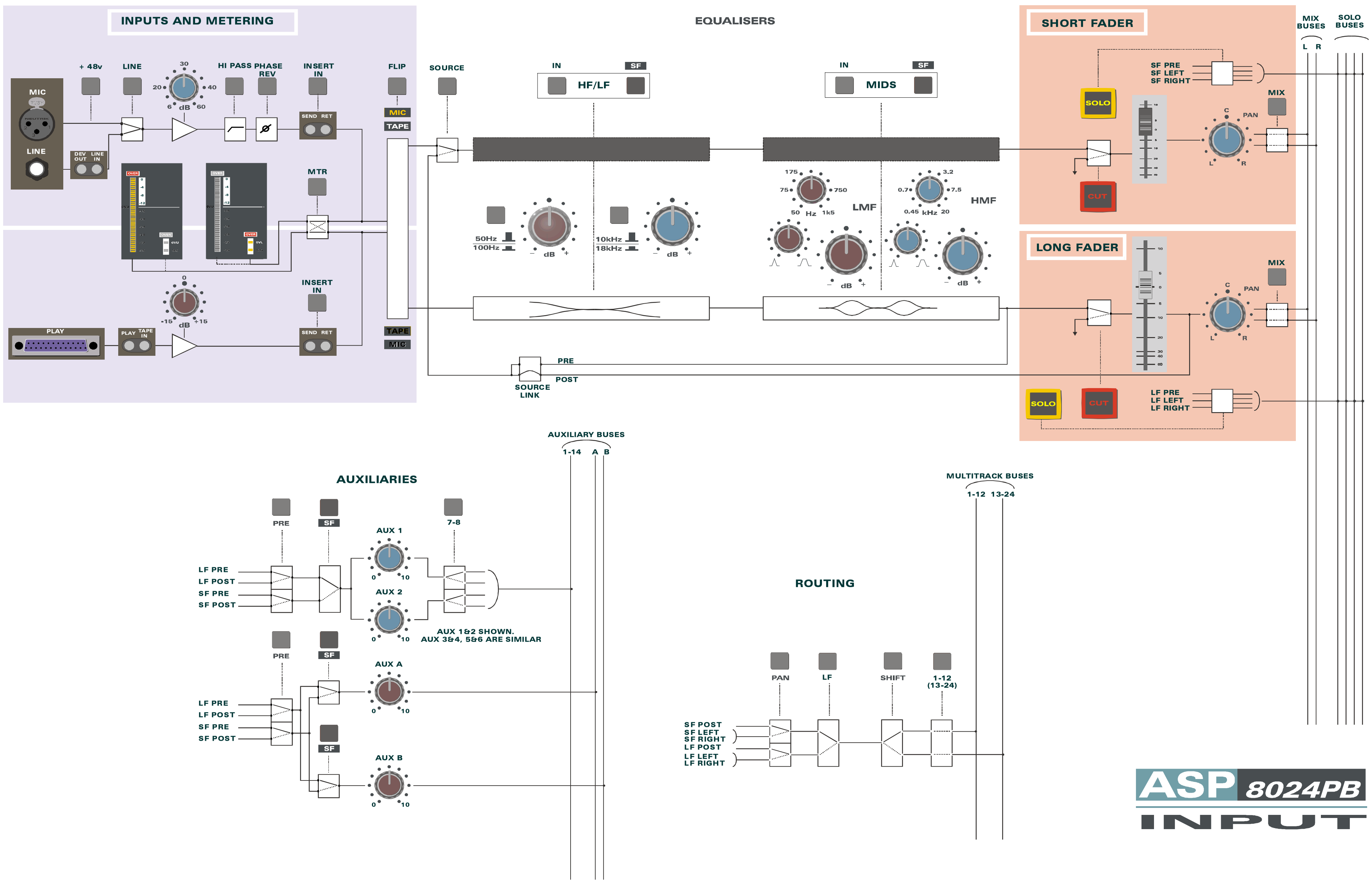
Foldback and Studio speakers

Oscillator

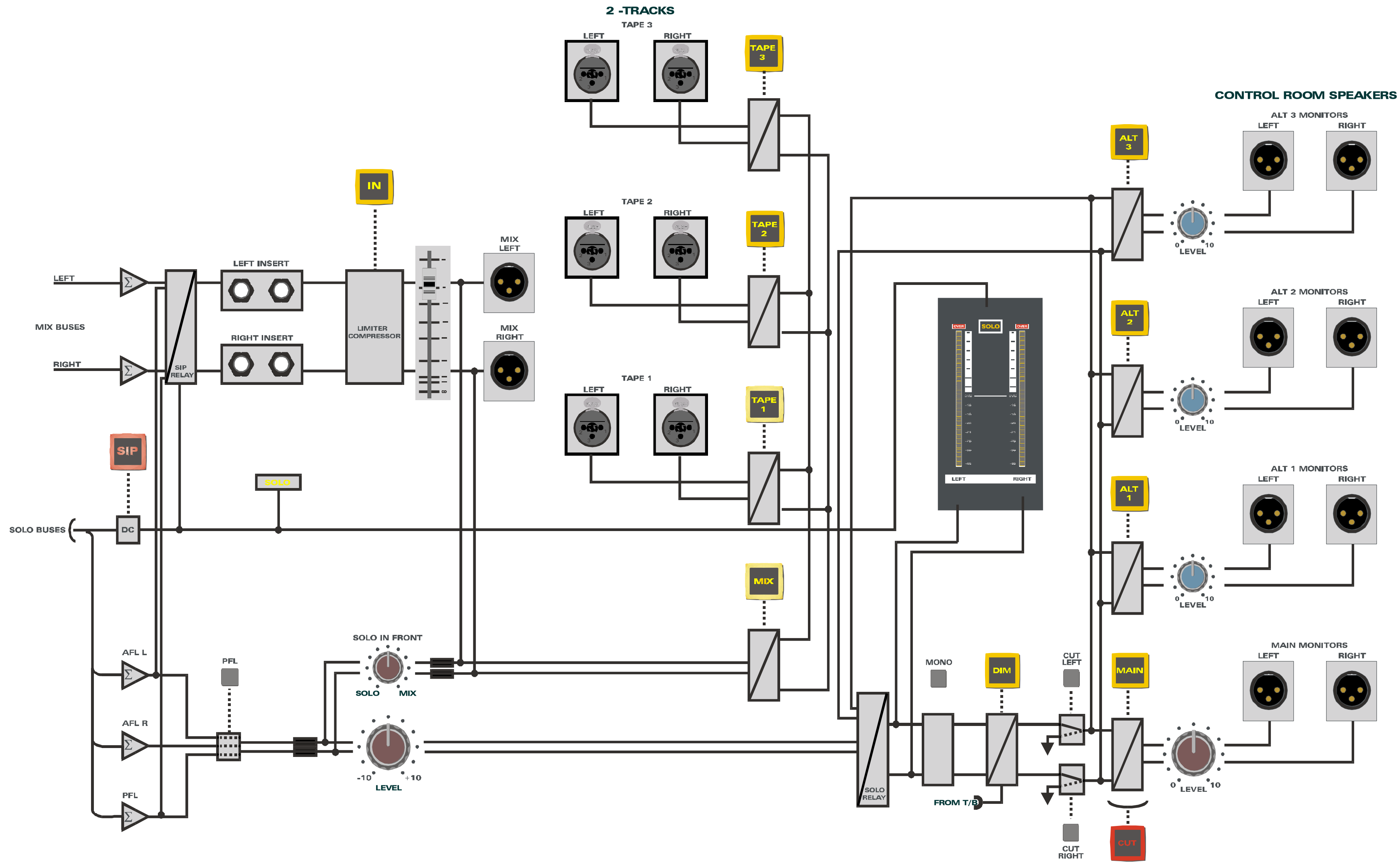
Talkback

ASP 8024 INPUT



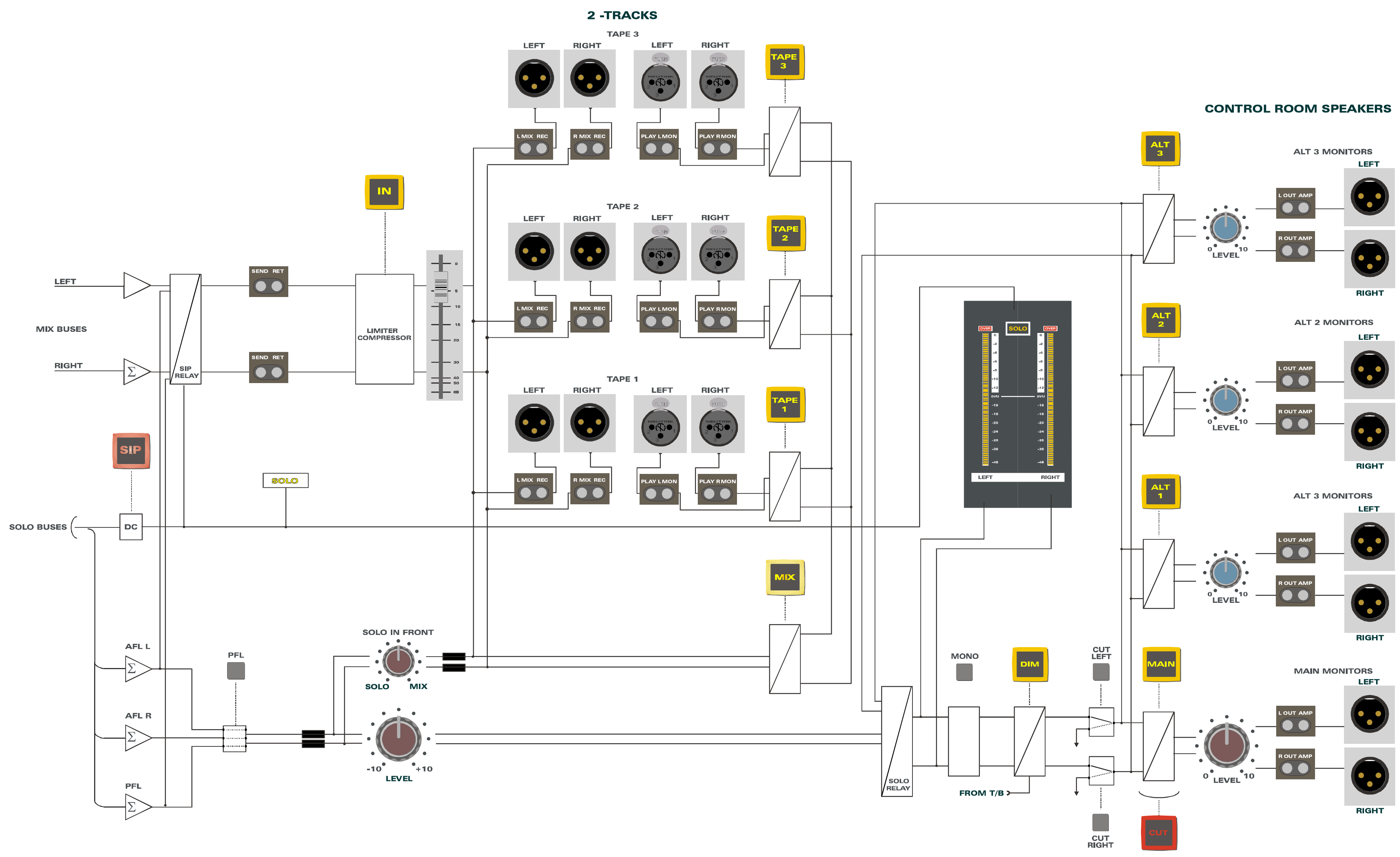


INPUTS AND METERING
EQUALISERS
SHORT FADER
LONG FADER
MIX BUSES
L R
SOLO BUSES
+ 48v
LINE
HI PASS PHASE REV
INSERT IN
FLIP
SOURCE
MIC
LINE
DEV OUT
LINE IN
SEND RET
MIC
TAPE
MTR
PLAY
PLAY TAPE IN
TAPE
MIC
IN
SF
HF/LF
MIDS
50Hz
100Hz
10kHz
18kHz
dB
LMF
HMF
175
750
75
50 Hz 1k5
3.2
7.5
0.7
0.45 kHz 20
SOLO
CUT
PAN
C
L
R
MIX
SF PRE
SF LEFT
SF RIGHT
LF PRE
LF LEFT
LF RIGHT
PRE
POST
SOURCE LINK
AUXILIARY BUSES
1-14 A B
AUXILIARIES
PRE
SF
AUX 1
AUX 2
AUX A
AUX B
7-8
LF PRE
LF POST
SF PRE
SF POST
AUX 1&2 SHOWN.
AUX 3&4, 5&6 ARE SIMILAR
MULTITRACK BUSES
1-12 13-24
ROUTING
PAN
LF
SHIFT
1-12
(13-24)
SF POST
SF LEFT
SF RIGHT
LF POST
LF LEFT
LF RIGHT
ASP 8024PB
INPUT



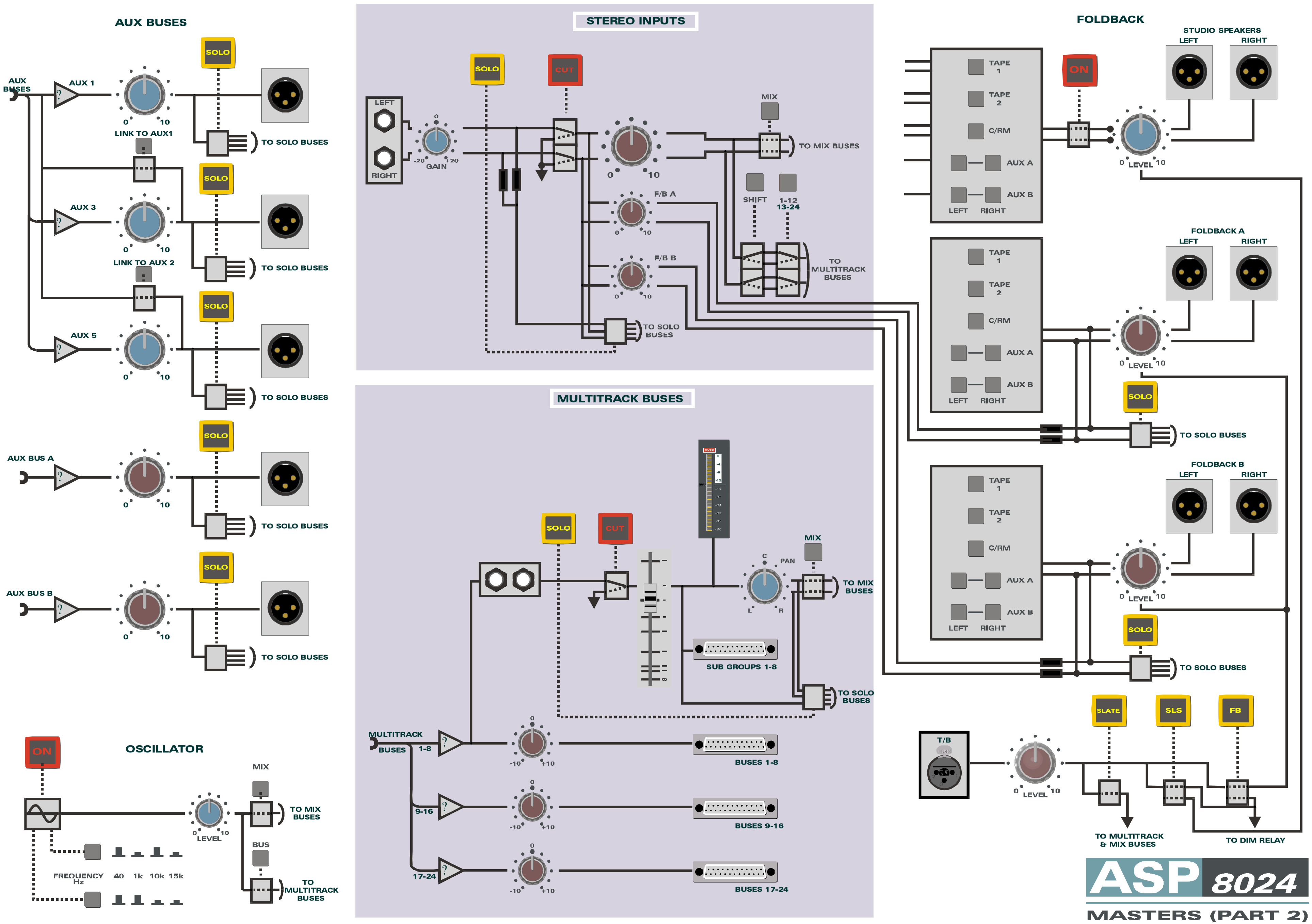
ASP 8024

MASTERS (PART 1)



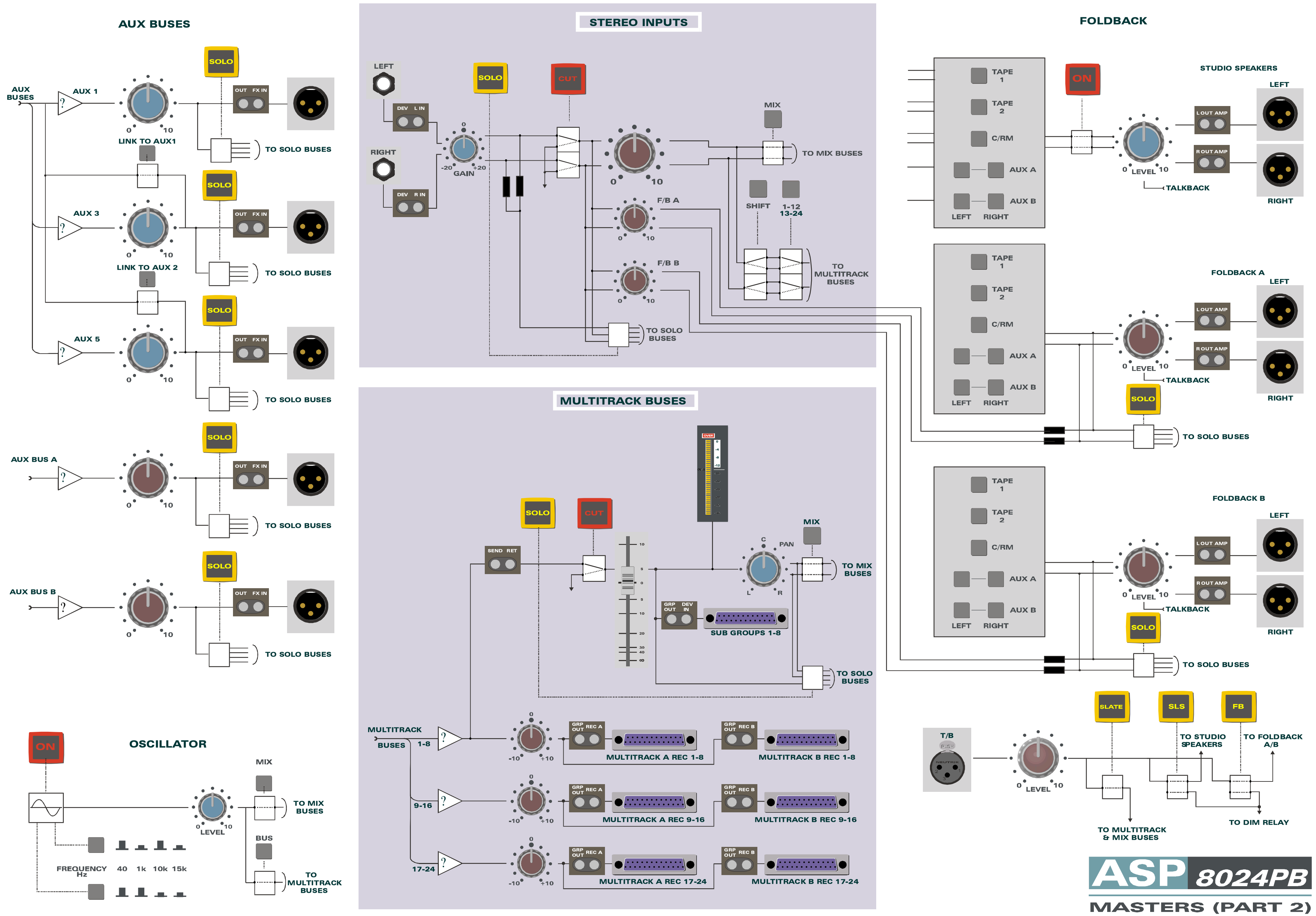
ASP 8024PB

MASTERS (PART 1)



ASP 8024

MASTERS (PART 2)



ASP 8024PB

MASTERS (PART 2)

WARRANTY

Your ASP8024 Console comes with a manufacturers warranty for one year from the date of despatch to the end user.

The warranty covers faults due to defective materials used in manufacture and faulty workmanship only.

During this warranty period Audient will repair or at its discretion replace the faulty unit provided it is returned carriage paid to an authorised Audient service centre.

We will not provide warranty repair if in our opinion the fault has resulted from unauthorised modification, misuse, negligence, act of God or accident.

We accept a liability to repair or replace your ASP8024 as described above. We do not accept any additional liability.

This warranty does not affect any legal rights you may have against the person who supplied this product – it is additional to those rights.