

THE SOUNDCRAFT GUIDE TO MIXING



Soundcraft

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STARTING OUT

A. What does a Mixer do?

No matter how sophisticated or expensive, all mixers carry out the same basic function - to blend and control the volume of a number of input signals, add effects and processing where required and route the resulting mix to the appropriate destination, which could be power amplifiers, the tracks of a recording device - or both. A mixer is the nerve centre of these sources, and therefore the most vital part of your audio system.

B. Guidelines in Choosing a Mixer

Audio mixers come in many different sizes and at all price levels, so it's little wonder that people are confused as to what type is actually needed for the job in hand. However there are several questions you can ask yourself that will help you narrow your search to the most appropriate models.

- What am I going to be using the mixer for - i.e. multitrack recording, live PA work or both?
- What is my budget?
- How many sound sources do I have? As a guideline your mixer needs to have at least as many inputs as sound sources. If you are likely to be buying more equipment in the future you should budget for extra inputs.
- What particular mixer facilities **must I have** for my application? i.e. plenty of EQ, auxiliaries, or Direct Outs for recording.
- How portable does the mixer need to be?
- Will I be doing any location work where there won't be any mains power available?
- Have I read the Soundcraft Guide to Mixing from cover to cover?

Once you can answer these questions satisfactorily you should have a fairly accurate specification for the mixer you need.

C. The Controls - A Description

This is where we get into the nitty-gritty of what controls and inputs/outputs you'll find on a typical mixer. For this example, we've used a Spirit SX. **If you are already familiar with what the controls on a standard mixer do, then it's OK to skip to section 2.** If you find a term particularly difficult, further explanation can be found in the Glossary (Section 8).

MONO INPUTS

A Mic In

Use this "XLR" input to connect your microphones or DI boxes.

For Mic Input Wiring Explanations see section 7.

B Line In

Use this connector for plugging in "Line Level" instruments such as keyboards, samplers or drum machines. It can also be used to accept the returns from multitrack tape machines and other recording media. The Line Input is not designed for microphones and although it may be used, will not provide optimum performance with them.

For Line Input wiring explanations see section 7.

C Insert Point

This is used to connect external signal processors such as compressors or limiters within the input module. The Insert Point allows external devices to be placed within the Input Path - see Fig. 1.1.

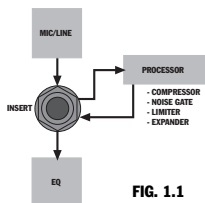
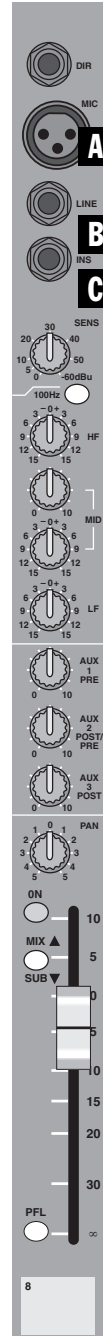


FIG. 1.1

See Section 2 and 3 for more detail on how to use processors, and Section 7 for information on wiring.



D Direct Out

This allows you to send audio direct from your channel out to a multitrack tape recorder, or to an effects unit when the channel requires its own special effect.

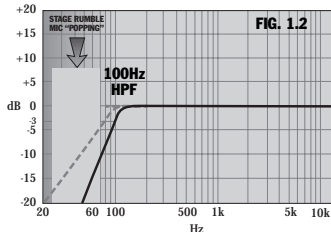
See sections 2 and 6 for more details on connections and studio techniques.

E Gain Control (Input Sensitivity)

Sets how much of the signal from the mic or line inputs is fed to the channel.

F HPF (High Pass Filter)

As the name suggests this switch cuts out the very lowest frequencies of a sound whilst allowing the higher frequencies to “Pass Through”. It’s particularly useful in live situations to reduce stage rumble or microphone ‘popping’, which can produce a muddy mix, or to ‘clean-up’ male vocals and filter out low frequency hum. Some manufacturers may also use the term “low - cut” filter to describe the HPF. See Fig. 1.2.



G EQ Section

Usually the most closely scrutinised part of any mixer, the equaliser section allows you to change the tone of the sound on each input. An EQ is normally split into “bands”, which control a range of frequencies, in a similar fashion to the treble and bass tone controls on your Hi-Fi. Indeed a simple “2 band” EQ is little more than an input treble and bass control. The more bands an EQ has the more sophisticated it is. SX has a 3 band EQ, with a separate control for the middle audio frequencies. This control is also “swept” which provides even more sophistication. Simply described, a sweep EQ allows you to choose the exact frequency to cut and boost, rather than having it chosen for you, as on normal “fixed” controls.

We will talk in more detail about EQ in section 3.

H Auxiliary Section

Typically, these controls have two functions: First, to control the levels of *effects* such as reverb from external effects units that have been added to the input signal, and second to create separate musician’s “*foldback*” mixes in the studio or on stage.

How to use auxiliaries, connecting them to external equipment and other applications are described in section 3.

I Pan (Panoramic Control)

This determines the position of the signal within the stereo mix image or may be used to route (send) the signal to particular GROUP outputs as selected by the ROUTING SWITCHES (see below).

J Solo (PFL and Solo in Place)

The PFL solo switch allows you to monitor an input signal independently of any other instruments that have been connected, which is useful for troubleshooting, or setting an instrument’s Input Preamp Gain and EQ setting.

Pre-Fade Listen (PFL) is a type of solo that allows you to monitor your sound BEFORE THE FADER. In other words when you move the input fader in PFL mode the level will not change, nor will you hear any effects. Because effects and volume are not a distraction, PFL solo is very useful for setting proper input preamp levels.

Some Soundcraft mixers use SOLO IN PLACE, which allows you to monitor signals after the fader in their true stereo image, and with any effects that have been added. This type of Solo is less good for level setting, but more useful in mixdown situations for auditioning sounds.

See section 3 - Setting Gain, for more information on using PFL.

K Mute/Channel On-Off Switch

This turns the channel on or off and is useful for isolating the channel when not in use or pre-setting channel levels which may not be needed until later, i.e: theatre scene-setting or support acts/performers.

L Fader

This determines the level of the input signal within the mix and provides a visible indication of channel level.

M Routing

By selecting the routing switches the input signal is sent to a choice of the mixer’s outputs - typically the main mix outs or the group outputs. The switches are used in conjunction with the PAN control to route the signal proportionately to the left or the right side of the mix or to odd/even groups/subs if PAN is turned fully left or right.

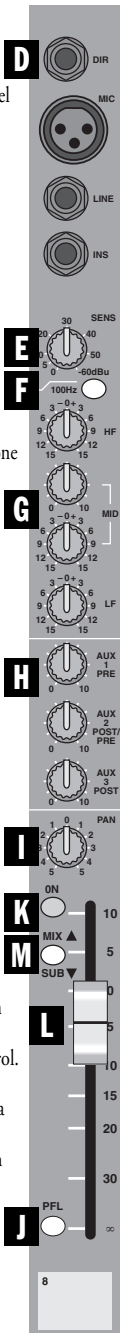


FIG. 1.3

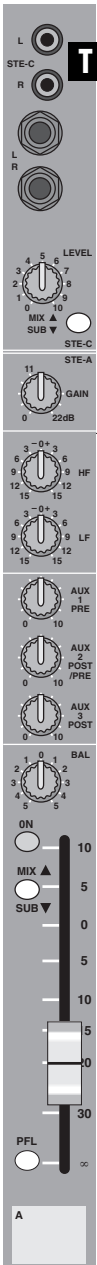


FIG. 1.4

STEREO INPUTS

Guitar amps and mic'd sound sources only provide you with mono signals. However keyboards, samplers, drum machines and other electronic media often provide true stereo outputs with separate left and right signals. Stereo Inputs on mixers simply allow you to connect both of these signals individually and control them from a single fader. Stereo inputs tend to incorporate fewer facilities than mono inputs as most keyboards are already equipped with plenty of internal effects and tone control options.

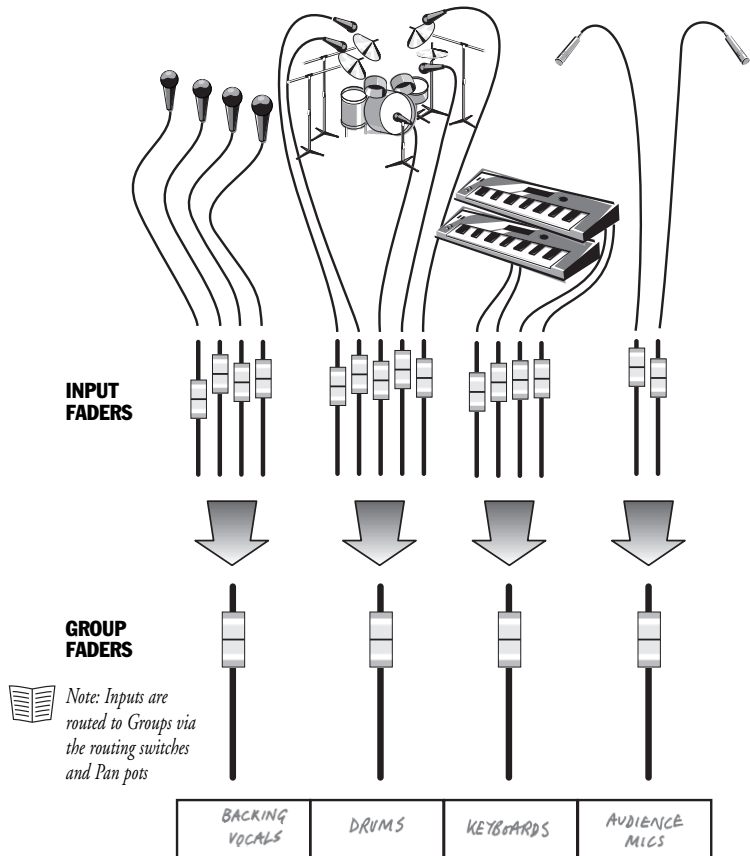
NB: Soundcraft jack stereo inputs default to Mono when the left input is used. RCA phono connectors do NOT have this option.



SUBGROUPS

These allow the logical assignment of groups of instruments or vocalists so that they may be controlled by just one pair of faders, or even a single fader, once individual instruments' relative levels have been balanced. They also act as additional outputs with separate volume/level controls – ideal for speaker fills or recording a number of instruments to one tape track.

FIG. 1.5



Note: Inputs are routed to Groups via the routing switches and Pan pots

THE MASTER SECTION

N Mix Outputs

Mix outputs provide left and right level control of the final stereo mix. Many consoles feature mix insert points too, allowing the connection of signal processors across the whole mix.

O Monitor “Engineer’s” / Control Room Outputs

These let you listen to any solo, mix, submix from a group, or the 2 Track tape return via an external amplifier and speakers, or the headphone socket.

P 2 Track Tape Returns

Allow you to connect the outputs of your cassette or DAT player and listen back to your completed masterwork. They may also be used for playing pre-show music at a gig using 2-Track to Mix switch (not shown in illustration).

Q Aux Masters

These govern the overall output levels from the auxiliary outputs and therefore the amount of signal going to an effects unit or a musician’s foldback mix.

R AFL

Allows monitoring of the actual signal leaving the Aux Masters.

S Meters

Normally they show mix output levels. When any Solo button is pressed, the meters automatically switch to show the solo level. They provide visual indication of what’s going on in your mixer.

T Stereo Returns (see Stereo Inputs earlier in this section)

These allow signals from external equipment, such as effects units, to be returned to the mixer and routed to the stereo Mix or Groups, without using up valuable input channels.

U +48v or Phantom Power

Some microphones, known as condenser mics, require battery power to operate. Alternatively the power may be provided by the console. This is known as ‘phantom power’ and runs at 48vDC. Simply press “Phantom Power” and any condenser mics connected will operate without the need for batteries.

More Information on Condenser Mics can be found in Section 3 - Mixing Techniques.

Further detail on mic wiring may be found in Section 7 - wiring.

V Headphones

Allow you to listen to your mix without annoying your neighbours or being distracted by ambient sounds.

That’s it, the basic features of your average mixing console. If you found it a little heavy going, don’t despair: it does get easier!

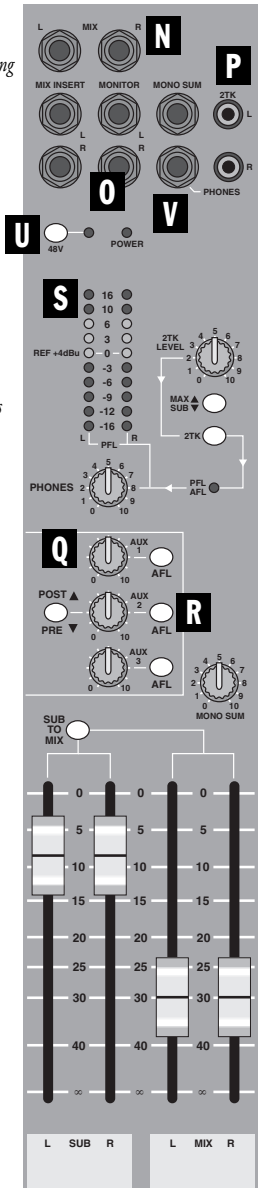


FIG. 1.5



Caution: DO NOT ACTIVATE A GLOBAL PHANTOM POWER SWITCH IF AN UNBALANCED SIGNAL SOURCES IS CONNECTED TO ANY MIC INPUT.
 Because of the voltage present on pins 2 and 3 of the XLR connector, you will damage your microphone/signal source.
 Always refer to your Mixer’s User Guide.

D. Signal Flow

Now the typical mixer features have been explained in detail it is important to understand how they form together. The route which a signal source takes through a mixer is normally shown using one of two devices: a **block diagram** or a **signal flow diagram**.

Both diagrams provide a 'visual' description of the key elements of the mixing console. They allow you to identify which components are used in the audio path and help the engineer to "troubleshoot" when signal sources don't appear to be doing what they should! In simple terms, they are electronic maps.

An example of a signal flow diagram is shown here. This is the most basic representation of console layout, showing a how a single sound source may pass through an input strip to the various other parts of the mixer.

Block diagrams are slightly more complex, showing more detail, electronic information, including the location of resistors and capacitors, and the structure of the entire console including bussing: **an example is shown on page 37**. Block diagrams also use a number of symbols to represent electronic elements. A few minutes spent understanding them some time during your journey through this booklet will most definitely pay-off in future mixing projects.

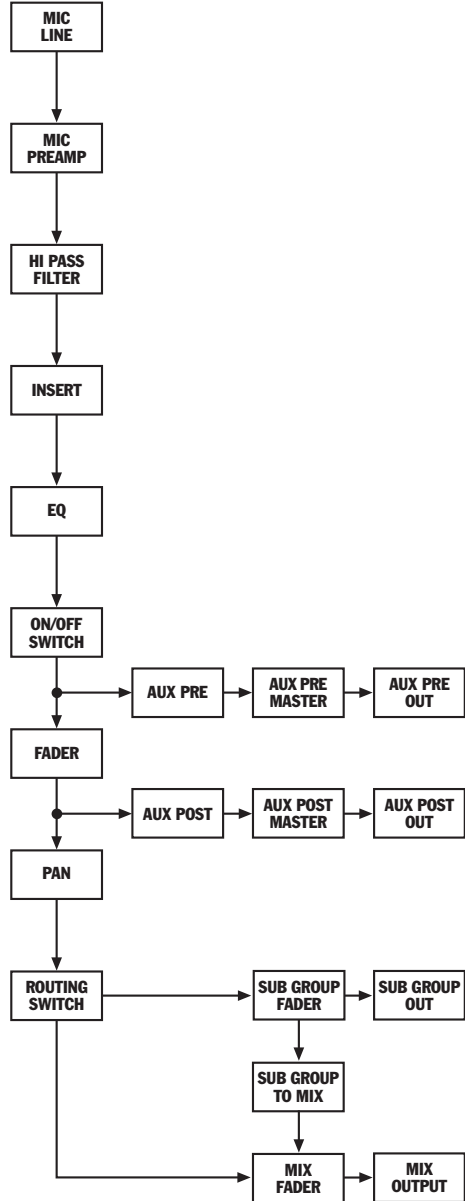


FIG. 1.6

A Typical Signal Flow Path

CONNECTING EQUIPMENT TO YOUR MIXER

As we explained in the last section, it is the job of the mixer to accept the various signal sources, set the levels and route those signals to the correct destination.

We'll now take a quick look at where to connect the 'peripheral' equipment that you will be using with your mixer. If you have already created your own set-ups successfully in the past, you should only need to skim this part.

A. Input Devices

Microphones

All microphones should be connected via each input's XLR connectors. Do *not* use line inputs.

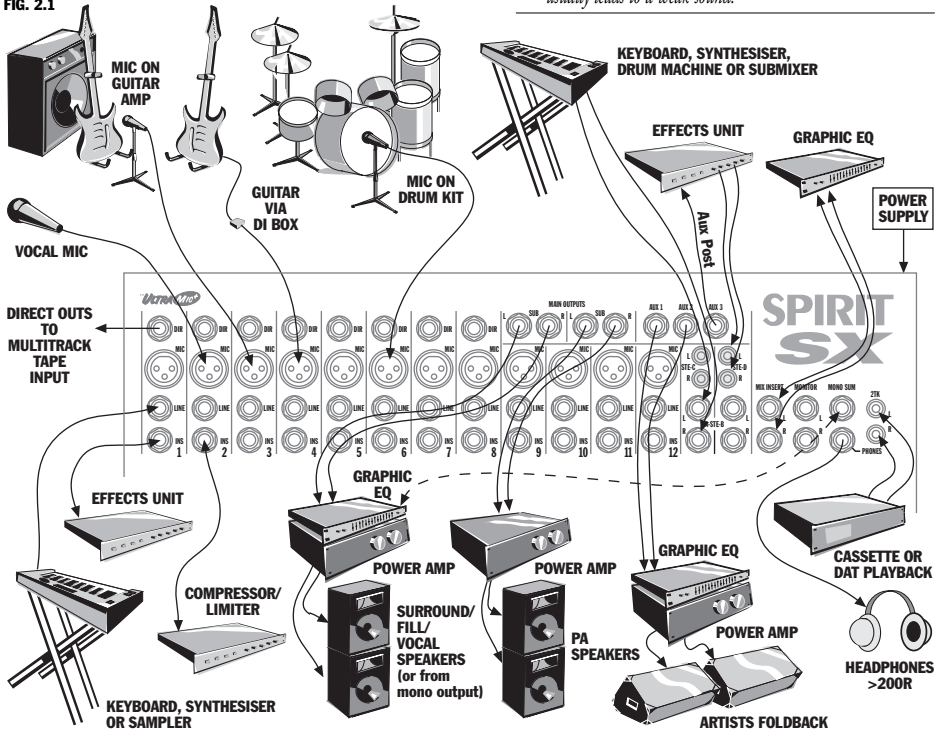
For more information on miking up individual instruments, refer to sections 4 and 6 - PA Mixing and In the Studio.

Direct Injection Box (DI Box)

- A DI Box allows you to connect a guitar or bass directly to the mixer's input, rather than miking up the instrument's amp/speaker. This technique is often preferred by musicians who require a "clean" sound. The best DI boxes are ACTIVE and require Phantom Power like condenser microphones. They should be connected to XLR mic inputs.

NB: Although electric guitars and basses may be connected to a mixer's line inputs without danger, the results will be far from ideal, because the IMPEDANCE of these instruments will not match up with typical line levels. Direct connection usually leads to a weak sound.

FIG. 2.1



Electronic Line Output Devices

- Keyboards, Drum Machines, CD Players, DAT Machines, Wireless Mic Receivers, all provide line level outputs, and should all be connected straight into the Mixer's Line Inputs. If some of your instruments are STEREO connect their left and right outputs to a spare stereo input. Alternatively connect to an adjacent pair of mono inputs and Pan the inputs hard left and right to create a stereo image.

B. Equipment Requiring Both Inputs and Outputs

External effects units

Connect the input of your effects unit marked "mono" to A POST FADER AUXILIARY OUTPUT. If you are uncertain, Post fader auxiliaries are coloured blue on Soundcraft mixers with the relevant channel aux pots usually marked "post". The left and right outputs from the effects unit should be connected to a pair of stereo returns, or stereo inputs if stereo returns are not available. If intensive EQ is required, use a pair of Mono Inputs. Remember, the effects signal is no different from any other audio signal – it still requires an input to the mixer.

See Section 3 Mixing Techniques or a detailed explanation of post fader auxiliaries.



NB: YOU DO NOT HAVE TO CONNECT UP BOTH THE LEFT AND RIGHT INPUTS OF YOUR EFFECTS UNIT TO SEPARATE AUXs. Most units only require "pseudo-stereo" operation and will mimic a stereo reverb or effect inside before providing a stereo output to the mixer's returns.

Signal Processors

Connect signal processors, such as compressors to the insert jack using a special insert 'Y' cable. This allows the signal to be sent and returned to the mixer using only one connector.

Refer to section 7 for wiring information.

It is also possible to connect the processor to the console without using the insert jacks by connecting an instrument direct to the processor first. However, the advantage of using processors in the mix/group or channel inserts is that any level changes made by the processor can be monitored by the mixers meters.



NB: A signal processor can be used in a channel to control one audio source, across a group to control a number of audio sources or across the entire mix.

Tape machines

Multitrack machines are used for initial track-laying in either studio or live recording situations. For more sophisticated work, a stand-alone machine offers better sound quality and greater versatility than a cassette multitracker. The new generation of digital multitracks are also very attractive, but analogue, open-reel multitracks are also capable of professional sounding results. Aim for a minimum of eight tracks if your budget will allow.

Mastering Machines

Your final mix should be recorded on the best quality machine that you can afford. A recording is only as good as the weakest link in the chain, and a good cassette machine is fine for demos, but for more serious work, consider a DAT machine or perhaps a second hand, open-reel 2-track.

C. Output Devices

Amps and Speakers (Monitor and FOH)

Studio Monitoring

A high-powered hi-fi amp of around 50 watts per channel is fine for home recording, but to ensure adequate head-room you should consider a well-specified rack mount amp. Similarly, a pair of accurate hi-fi speakers will do the job, but for more serious work we would recommend purpose-designed nearfield monitors. Always remember that no matter how good the recording or performance, a poor monitoring set-up will not allow you to make qualitative judgements about the mix.

Headphones

When choosing headphones for monitoring, you'll obviously want a pair that give the best sound reproduction for the price. But, bear in mind that in order for you to fully concentrate on the mix, the headphones should exclude outside noise - therefore open-back designs will be of little use.

Furthermore, you could be wearing the headphones for several hours at a stretch so comfort is essential.



NB: Make Sure that the IMPEDANCE of your headphones matches the specification of your mixer.

PA Work

PA work requires high-powered, rugged, and honestly specified amps and FOH (Front of House) speakers. The power rating of the system will depend on the size of venues you will be playing. See PA Mixing, Section 4, for more information.

MIXING TECHNIQUES

A. Choosing the Right Microphone

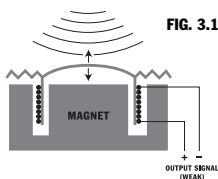
Microphone Types

The choice of microphone depends on the application that the microphone will be used for and individual preference. However, broadly speaking microphones fall into two main types:

Dynamic Microphone -

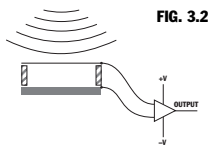
- A robust design which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

Dynamic mics are relatively inexpensive, rugged and require no electrical power to operate. They are ideal for all-round high sound pressure levels (SPL) and tend to be used for live applications. However, they are not as sensitive to high frequencies as condenser types.



Condenser Microphone -

- A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. They need power to operate - the most common source being +48v DC PHANTOM POWER. Condenser mics are very sensitive to distant sounds and high frequencies. Because of this sensitivity they are often used in studio recording situations.



Microphone Pick-up Patterns

A pick-up (Polar) pattern refers to the area(s) from which a microphone "picks up" its sound. It is important to choose the right pattern for your application, or you may pick up sounds from areas you don't want or lose sound information you need.

Omni Pattern

The most basic type of microphone pattern.

- A 360° polar response which picks up sound equally in all directions.

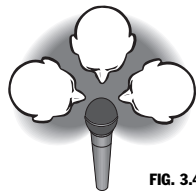
This pattern is ideal for picking up groups of vocals, audiences, ambient sounds but is most susceptible to feedback.



Cardioid Pattern

- The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front.

Used for most basic recording or in any situation where sound has to be picked up from mainly one direction. Dynamic cardioid mics are mostly used for live applications because they help reduce unwanted spill from other instruments, thus reducing the risk of feedback.



Hyper-cardioid

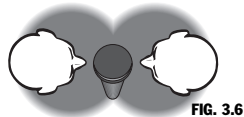
- Similar to a cardioid pattern but with greater directionality. Used for live vocal microphones because it provides the greatest protection from unwanted spill and feedback.



Figure of Eight

- Sound is picked up from the front and back but not from the sides.

This pattern is used mainly in studios for picking up two 'harmony' vocalists, or solo vocalists who require some room ambience.



N.B. +48v Phantom power is used to charge the diaphragm and plate. It also supplies a small amplifier which boosts the small voltages generated by diaphragm movements.

B. Setting Up a Basic Mix

Setting the Gain

Input gain is designed to take an audio signal, and adjust it to the level which the mixer understands.

All audio circuits, mixers included, produce a low level of electronic noise or hiss, and while this can be made very low by careful design, it can never be completely eliminated. It is also true that any audio circuit can be driven into distortion if the input is too high in level; hence care has to be taken when setting the input level so as to

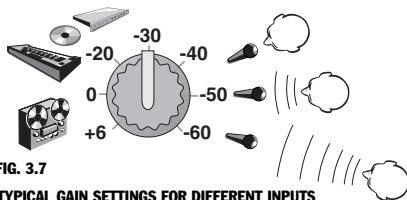


FIG. 3.7

TYPICAL GAIN SETTINGS FOR DIFFERENT INPUTS

preserve the best possible sound quality. Ideally the input signal should be as high in level as possible while still leaving a margin of safety to prevent distortion on loud sections. This will ensure that the signal is large enough to render the background noise insignificant, whilst keeping the signal clean. The remaining safety margin is known as *Headroom*.

To set the gain on the mixer;

- Press the PFL/Solo switch on the relevant input.
- Adjust gain/input sensitivity until meters read within the yellow ('3' to '6' on meter scale). This allows for the extra 10dB of gain that is available on Soundcraft input faders.
- Release PFL/Solo.
- Repeat for all other inputs.



NB: EQ affects gain settings. If you adjust the EQ you will need to re-check your gain level using the above method.

Once you have optimised the gain your mixer will give the best possible signal quality with the minimum of noise and distortion.

Balancing Fader Levels

Faders allow you to make fine adjustments to your sounds and act as a visual indication of the overall mix levels.

It is important to keep your input faders around the '0' mark for greater control. This is because fader scales are typically logarithmic and not linear, so if your fader position is near the bottom of its travel then even a small movement will lead to huge leaps in level. Similarly try not to have your fader at the top of its travel because this will leave you no room to further boost the signal.

See diagram below.

Balancing Output Levels

Master Outputs

Set your master outputs to '0' on the scale. There are two reasons for this:

- 1 You have the maximum fader travel for fading out your mix.
- 2 If your faders are set below '0' you will not be getting the full benefit from the meters because you will only be using the first few LEDs on the meter scale.



NB: Your mixer is not an amplifier. So the master output faders should be set to maximum ('0' on scale). If extra output is required, then turn up your amplifier.

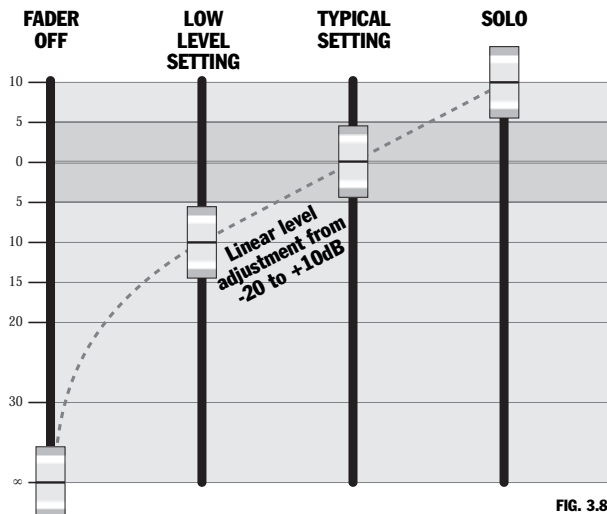


FIG. 3.8

C Using the Mixer's EQ

Equalisation is useful for making both corrective and creative changes to a sound, but it needs to be used with care. Corrective applications include making tonal changes to compensate for imperfect room acoustics, budget microphones or inaccurate loudspeaker systems. While every effort should be made to get the sound right at source, this is less easily achieved live than in the more controlled conditions of the recording studio. Indeed, the use of equalisation is often the only way to reach a workable compromise in live situations.

Creative applications, on the other hand, are equally as valid in the recording studio as they are live, and an equaliser with a swept midrange control is infinitely more versatile than one that has simple high and low controls. The only rule of creative equalisation is - 'If it sounds good, it is good!'

Fixed EQ

Most people will be familiar with the operation of high and low frequency controls; they work in a similar manner to the tone controls on a domestic stereo system.

In the centre position the controls have no effect, but rotate them clockwise and they will provide boost, or rotate them anticlockwise and they provide cut. Despite their apparent simplicity, however, high and low controls should be used with caution as overuse can make things worse. Adding a small amount of high or low boost should be enough to add a touch of brightness or warmth to a sound, but a quarter of a turn should be sufficient, especially where the low control is concerned.

The drawback with fixed controls often lies in the fact that you may want to boost just a particular sound such as the punch of a bass drum or the ring of a cymbal, whereas a fixed control influences a relatively large section of the audio spectrum. Apply too much bass boost and you could find the bass guitar, bass drum and any other bass sounds take on a flabby, uncontrolled characteristic which makes the mix sound muddy and badly defined. This is because sounds occupying the lower mid part of the spectrum are also affected. Similarly, use too much top boost and the sound becomes edgy with any noise or tape hiss being emphasised quite considerably.

In a PA situation, excessive EQ boost in any part of the audio spectrum will increase the risk of acoustic feedback via the vocal microphones.

THE FREQUENCY RANGE OF DIFFERENT INSTRUMENTS AND WHICH EQ BANDS AFFECT THEM

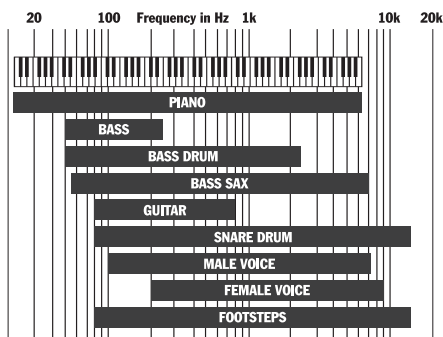
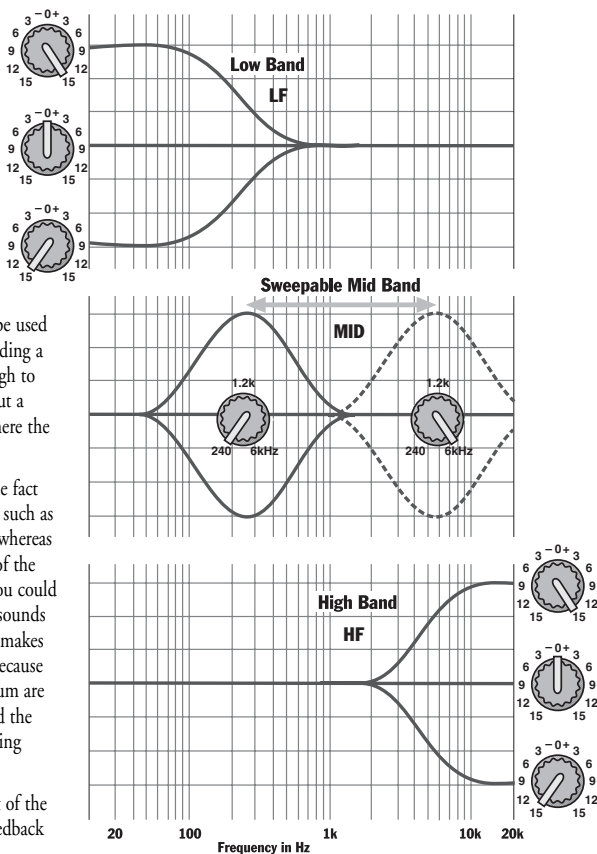


FIG. 3.9



Bearing the above points in mind, the best approach is to use small amounts of boost, especially when working live. EQ cut, on the other hand, causes far fewer problems, and *rather than boost a particular sound it is frequently more rewarding to apply cut in whichever part of the audio spectrum that appears to be overpowering*. In this application, the sweep mid control is also very effective.

Using a sweep-mid equaliser

Like the high and low controls, the sweep mid can provide either cut or boost, but its strength comes from the fact that it can be 'tuned' into the specific part of the audio spectrum that needs treatment. Like the high and low controls, it is more forgiving if used to cut rather than to boost. However, when first tuning in the mid control, it helps to set it to full boost, so that when the frequency control is adjusted, the effect is most apparent. This is true even if the final EQ setting requires cut rather than boost.

Procedure

Below is a simple way of eliminating unwanted sounds:



Caution: when adjusting EQ, there is a danger of feedback which can cause damage to your speakers. You may need to reduce your levels to compensate.

- Increase sweep-EQ gain.
- Sweep the frequency pot until the aspect of the sound you wish to modify becomes as pronounced as possible. This should only take a few seconds.
- The cut/boost control is now changed from its full boost position to cut. The exact amount of cut required can be decided by listening to the sound while making adjustments.
- Even a small amount of cut at the right frequency will clean up the sound to a surprising degree.

Other sounds may benefit from a little boost, one example being the electric guitar which often needs a little extra bite to help it cut through the mix. Again, turn to full boost and use the frequency control to pick out the area where the sound needs help. Then it's a simple matter of turning the boost down to a more modest level and assessing the results by ear.

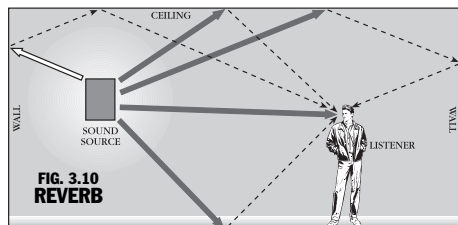
D. Using Effects Units

The Different Types

The problem with mixing 'dry' (using no effects) within a live or recording environment is that the results can often sound boring and lacking in colour. This is especially the case as most of us are used to listening to highly polished CDs at home. These productions are actually achieved by using effects which electronically produce certain atmospheres. The different types of effects that can be used are explained below;

Reverb

Reverberation is the most commonly used studio effect, and also the most necessary. Western music is invariably performed indoors where a degree of room reverberation is part of the sound. Conversely, most pop music is recorded in a relatively small, dry-sounding studio, so artificial reverberation has to be added to create a sense of space and reality. Reverberation is created naturally when a sound is reflected and re-reflected from the surfaces within a room, hall or other large structure. See fig. 10.



Delay

Often used to make a sound 'thicker' by taking the original sound, delaying it, then mixing it back with the original sound. This short delay added to the original sound has the effect of doubling the signal.

Echo

A popular effect that was used extensively on guitars and vocals in the 60s and 70s. It is not used on vocals so much nowadays, but quite effective on guitars and keyboards. A neat trick is to set the echo delay time so that the repeats coincide with the tempo of the song.

Chorus & Flanging

Both chorus and flangers are based on a short delay, combined with pitch modulation to create the effect of two or more instruments playing the same part. Flanging also employs feedback and is a much stronger effect. Both these

treatments work well on synth pad sounds such as strings and are best used in stereo where they create a sense of movement as well as width.

Pitch Shifters

These change the pitch of the original signal, usually by up to one octave in either direction and sometimes by two. Small pitch shifts are useful for creating de-tuning or doubling effects. Which can make a single voice or instrument sound like two or three, while larger shifts can be used to create octaves or parallel harmonies.

NB: For useful effects settings with different instruments refer to Section 6 'In the Studio'.

Setting up an effects loop

- Set effect unit to full 'wet' signal
- Connect your effect units as per Section 2, Input Devices.
- On the relevant input channel, set the post fade aux to maximum
- Select AFL on your aux master
- Set aux master level so that the meters read '0'
- Adjust input level on effects unit until 'effects meters reads '0' (nominal)

NB: You can now use the mixer AFL meters to monitor effects unit levels as both meters have been calibrated.

- Release aux master AFL and select effects return PFL

NB: If you are using a simple stereo input with no PFL, adjust input gain for required effect.

- Adjust effects return input gain until meters read around '0'.
- De-select PFL and adjust effects return fader level for required effect level.

NB: The original 'dry' signal is mixed with the effects 'wet' signal.

Pre- and Post-fade Auxiliaries

Pre-Fade

Pre-fade auxiliaries are independent of the fader so that the amount of effect will not change with new fader levels. This means you will still hear the effect even when the fader is at the bottom of its travel.

Post-Fade

It is important to use post fade auxiliary sends for effects units. This is because post fade auxiliaries 'follow' the input fader so that when input level changes the amount of effect remains proportional to the new input level.

NB: Effects Return Aux Post Control must be set to minimum or feedback will occur

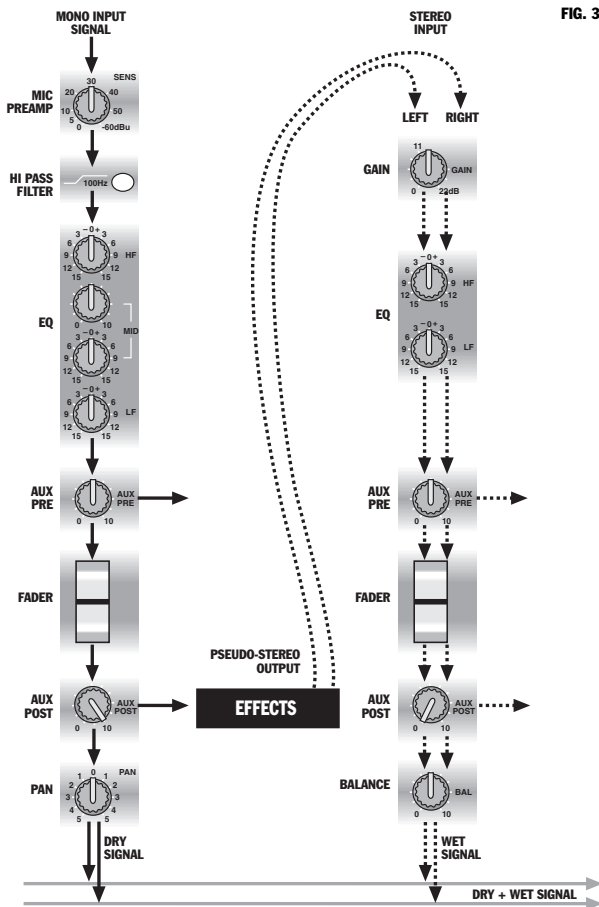


FIG. 3.11

E. Using Signal Processors

The Difference between Signal Processors and Effects

Unlike effects, which are creative in nature, signal processors are used to control and manipulate sounds to achieve the best audio quality performances and recordings.

Effects and signal processors should never be confused. Whereas effects are “mixed” with an input to provide a combined sound, signal processors alter an input, group or mix signal completely. The signal is actually taken out of the mixer entirely, “processed” and returned in its altered state, in series with the original audio signal.

For this reason signal processors should be connected using Insert Points and not the Auxiliary Send and Return Loop (effects loop).



NB: Effects can be connected to inserts if necessary, but then the proportion of the effect in the signal is governed solely by the effects unit mix control.

The Different Types of Signal Processors

Broadly speaking, there are 5 different types of signal processor in common use:

Graphic Equalisers

Graphic Equalisers work by splitting the sound spectrum into narrow, adjacent frequency bands and giving each band its own cut/boost slider. The term Graphic comes about because the position or ‘curve’ of the sliders gives a graphic representation of the way in which the settings affect the audio frequency range.

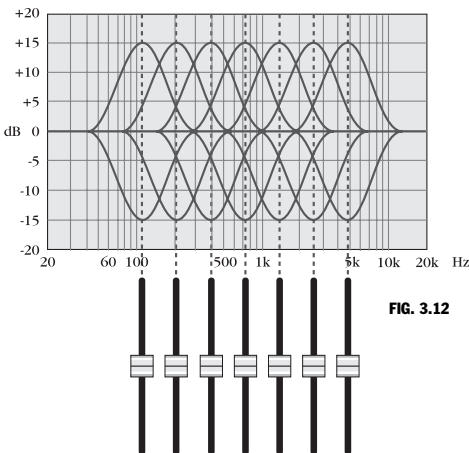


FIG. 3.12

Graphic Equalisers are most often used to process the mix at live venues by notching out troublesome frequencies that may be causing feedback. They may also be used to enhance a mix at a poor sounding venue. In recording they are used to create “flat” listening environments.

For more detail on venue acoustics go to section 4 - PA Mixing.

Parametric Equalisers

These are similar to the EQ found on an input channel but may include more bands and additional bandwidth (Q) controls which define how many frequencies in the band are affected.

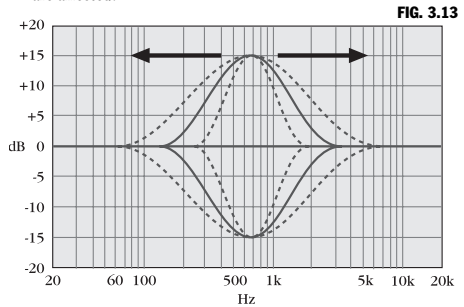


FIG. 3.13

They are most often used to provide additional creative control over an input signal when a mixer’s EQ is not sufficient.

Gates

A gate is designed to shut down the audio signal path when the input signal falls below a threshold set by the user. It may be used to clean-up any signal that has pauses in it. For example gates are widely used to prevent ‘spill’ between adjacent mics on a multi-mic’d drum kit where, say, a tom-tom mic may pick up the snare drum.

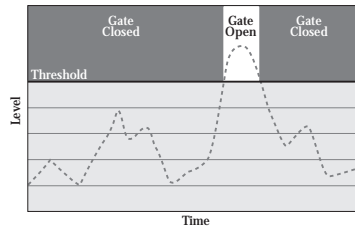


FIG. 3.14

Expanders

Expanders accomplish much the same task as gates, though they are more like compressors in reverse. Compressors affect the gain of signals exceeding the threshold, while expanders act on signals falling below the threshold. A gate will close completely when the signal falls below its

threshold, but an expander works like an automatic mixing engineer who pulls down the signal when the signal falls below the threshold; the more it falls below the threshold - the more he pulls down the fader.

Expanders are most often use in Studio recording to provide the best mix signal to noise ratio when producing final masters.

Compressor/Limiters

A compressor reduces the difference between the loudest and quietest parts of a performance. It works on a threshold system where signals exceeding the threshold are processed and those falling below it pass through unchanged. When a signal exceeds the threshold the compressor automatically reduces the gain. How much gain reduction is applied depends on the 'compression ratio' which on most compressors is variable: the higher the ratio, the stronger the compression. Very high ratios cause the compressor to act as a limiter where the input signal is prevented from ever exceeding the threshold.

Compressors are the most commonly used processor and are particularly popular for maintaining constant vocal and bass guitar levels live and in the studio. This is because, out of all instruments, singers tend to vary their levels the most. Compressors help to achieve the much sought-after tight, "punchy" sound.

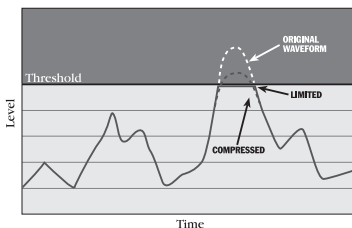


FIG. 3.15

Setting up a Signal Processor

- Connect your processor to the relevant mixer insert jack (mono, group or mix insert), using an insert 'Y' lead.

Refer to section 7 for wiring information

- Set your processor to unity gain (x1), i.e. no additional gain.
- Make your adjustments on your signal processor
- Beware that your processor settings may alter your mixer input output levels. Re-adjust levels to '0' on meters, if necessary.



NB: Remember a signal processor can be used in a channel to control one audio source, in a group to control a number of audio sources, or to control the entire mix.

F. Creating a Foldback/Monitor Mix

Performers usually require their own mix independent from the main/engineer's mix. This is because to achieve the optimum performance they need to hear themselves above other voices or instruments. This performer's mix is known as a foldback/monitor mix.

The procedure is as follows;

- Set the **pre-fade** aux to maximum on the relevant performers input channel.
- Select AFL on your aux master.
- Set aux master level so meters read '0'.
- Create a foldback mix for the performer by setting the pre-fade aux levels on the other performer's input channels.
- Release aux master AFL.



NB: It is typical that the performers' own vocals/instruments will be two thirds louder than any other sources in their own monitor mix.

Each performer may require a separate monitor mix/auxiliary output.

NOTE: Pre-fade rather than post-fade auxiliaries must be used. This is because they are independent of the input faders. If post-fade auxiliaries are used, then foldback mix levels will alter with every input fader change made by the FOH engineer. This will annoy the band and may lead to feedback which can damage speakers and headphones.

Now that you know how to connect and set up different elements of your system let's look at some real-world examples of systems in use.

PA MIXING

A. A Typical Live Performance

Introduction

There are so many different types of 'live' scenarios that it would be almost impossible for us to describe each one in a book of these modest proportions. Instead, our 'typical live gig' is represented by a small band, whose set-up is shown in the "Mixing Live" diagram.

Microphones

Most of the microphones used in live applications are dynamic cardioids because they are tough, produce an intelligible sound and their directional response helps prevent spill or feedback. Dynamic microphones can handle anything from drums to vocals. However, condenser types, with their greater sensitivity to high frequencies are invariably used for jobs such as overhead pick-up on a drum kit or mic'ing acoustic instruments.

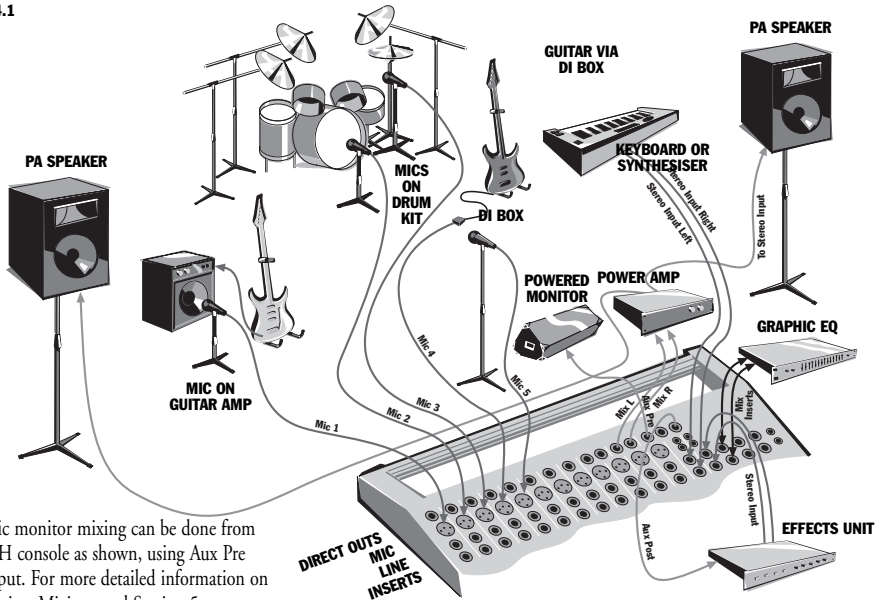
Cables and Connections

Interference and hum can be avoided! A few minutes spent checking cable runs and connectors pays dividends.

- A balanced audio connection provides low noise operation by cancelling out any interference in a signal. It does this by using a 2-conductor mic cable surrounded by a shield. Any interference picked up will be of the same polarity on the two conductors and is therefore rejected by the mic input's Differential Amplifier.
- Don't skimp on interconnecting cables - always buy the best that you can afford. Make sure that all connections are sound and keep cable runs as short as is practicable.
- A multicore cable and stage box will keep trailing cables to a minimum and presents a tidy and practical approach.
- If your mixer has a separate power supply unit, keep it well away from the console.
- Where signal and mains cables must cross, make sure they're at 90° to each other. This will help reduce the risk of hum and noise.
- If the venue has a three-phase supply, don't share the same phase as lighting controllers.

MIXING LIVE

FIG. 4.1



- Basic monitor mixing can be done from FOH console as shown, using Aux Pre output. For more detailed information on Monitor Mixing, read Section 5.

- It is dangerous to lift the mains earth when trying to eliminate hum. You can isolate hum by lifting the appropriate audio signal shield.
- When using wireless mics, set the receiver on stage and run back to the console at balanced mic level. This will help avoid interference from digital sources and lighting controllers.
- Keep unbalanced 'insert' leads away from mains and keep them short - no longer than about 2 metres.
- Be sparing on the use of artificial reverb. Most venues are too reverberant anyway, and excessive reverb will ruin the intelligibility of the vocal performance.
- Do not use reverb on low frequency sound sources such as bass, kick drums and tom toms.
- Keep backline amp levels down: let the mic and mixer do the work!
- Always leave a little gain in hand so you can wind up the level slightly as the show progresses.
- Putting high levels of bass guitar or kick drum through a small PA can overload the system and distort vocal quality. Try rolling off some of the low bass, you'll get a higher subjective sound level without overload.

Connecting External Effects and Processors

We talked about Effects and Processors in Sections 2 and 3, so you're now aware of their functions and applications. Effects units are best connected via the console's Auxiliary Send and Return Loop (sometimes known as the Effects Send and Return Loop) or the Insert Point. When used in the Aux Send system, the dry signal level should be turned off on the effects unit, but when used via Insert Points (for guidance on how to wire a jack for use with Insert Points, see Section 6), the dry/effects balance must be set on the effects unit itself. Processors treat the whole of the incoming signal and therefore may only be used via console Insert Points or directly 'in-line' with a signal: they cannot be used in the Aux Send/Return loop system.

Setting Up

- Position the mixing console so that you can hear the on-stage performance as the audience will hear it. Ensure that you have a clear view of the performers.
- After setting up, switch the power amps on last to prevent any thumps occurring when effects or instruments are powered up. Ensure the console's master gain is down before you switch on the amplifiers.
- Don't set up the vocal mic directly in front of the drum kit or a guitar stack.
- Make sure the speakers aren't obstructed by the audience and that the majority of the sound is being directed towards the audience, not towards the rear or side walls.
- Set up the vocal levels first - it's no use getting a great drum sound if the vocals feed back before they can even be heard.
- Keep the vocals panned towards the centre of the mix. Not only will this sound more natural, but it will allow the greatest vocal level before feedback or distortion occurs.

Ringling Out: Nulling Room Acoustics



Caution: Ringling out can cause howl around which can damage speakers, so use care when adjusting levels.

As experienced engineers will tell you, there's no such thing as a perfect venue. To help tailor the sound to the room acoustics, insert a Graphic Equalizer into the console's mix insert jacks which are effectively between the mixer and the power amp.

'Ringling Out' the system prior to the sound check will help reduce troublesome feedback. To Ring Out, follow this procedure:

- 1 Set all graphic EQ controls to centre (0).
- 2 Turn up amp volume until feedback is just beginning to 'ring'.
- 3 Turn back the amp volume slightly to prevent accidental feedback.
- 4 Starting from the left, adjust the first graphic EQ frequency gain control to 'max': if the system doesn't feedback, then this is not a problem frequency. Return this gain control to centre position.
If the system feeds back, reduce the EQ gain by the same amount you boosted to get feedback.
- 5 Repeat this procedure for all graphic EQ frequencies.

Setting the Mix

- Turn down the amplifier gain before the system is first switched on. This will avoid unwelcome howls of feedback and can prevent loudspeaker damage due to switch-on transients.
- Set all the channel EQs to their flat or neutral position and optimize the input gain control setting for each channel in turn using PFLs.
- If low frequency background noise is a problem, switch in the High Pass Filter on each of the microphone channels being used, except on low frequency sound sources such as basses and kick drums.
- Ring out the system as described above, with the vocal mics open, and notch out any obvious trouble spots.
- Establish the maximum working level for the lead vocal mic so as not to incur feedback and then work a little below this level to allow a margin of safety. Again, see the notes on ringing out the system.
- Set up the backing vocal mics and check that there is no feedback problem when both the backing vocal and lead vocal mics are on. If there is, reduce the master gain setting until the feedback disappears.
- Now the instrument and direct line inputs can be balanced relative to the vocals. Start with drums and work through to the bass and rhythm instruments.
- Test out any effects units connected to the system and establish the correct balance of dry and effected sound.

Avoiding Feedback

- Turn down or mute any mics not in use. This reduces the risk of feedback and avoids the back line being picked up.
- If feedback is a real problem, consider moving the main PA speakers away from the mics a little. Also check the back of the stage, because if the wall is acoustically reflective, some sound from the room will be reflected back into the mics increasing the risk of feedback.
- Avoid excessive use of boosted EQ as this can encourage feedback and may also spoil the basic character of the sound. Consider it an aid to fine tuning rather than as a means of making radical changes.
- The use of stage monitors will also worsen the feedback situation so run these at the lowest volume that the performers can comfortably work with. Position the cabinets so as to allow as little direct sound as possible to enter the vocal microphones. If possible, use a graphic EQ on each monitor.



NB: Remember, people soak up sound! The perfect mix achieved in an empty venue will have to be tweaked when the crowds arrive. Sound waves are also affected by heat and humidity.

B. Larger Performances

Although the example shown in the 'Mixing Live' diagram shown at the beginning of this section is of a small band, the principles are the same no matter the size of the live performance or venue. However, for larger PAs additional speakers, monitors, effects and processors may be required as well as slightly different positioning for each of these pieces of equipment. These additional requirements are outlined below:

Medium Sized Venues

The console used will require more input channels. For example it is likely you will want to mic up all of the drums, and there are also likely to be more instruments, backing singers and sound sources in general.

More monitor sends will also be required - a single monitor will not be enough for larger bands. The bass and drums will require a monitor between them. The vocalists will want a monitor each so they can hear themselves above the band.

More speaker outputs may be needed in larger venues so that all the audience can be reached, without there being "holes" in the amplified audio signal. It may be necessary to record the event. This will require additional level controlled stereo outputs or direct outs if a multitrack is being used.



NB: For simplicity, these diagrams do NOT show any outboard equipment.

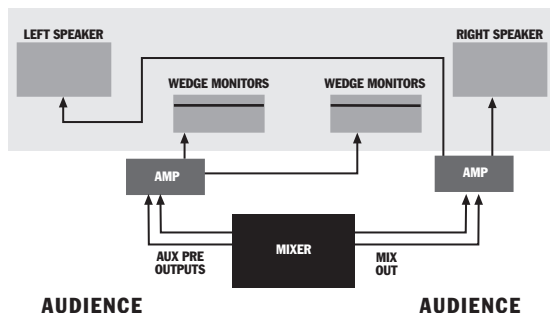
Large Sized Venues

Large venues will require a separate "Front of House" (FOH) console for the audience mix and a Monitor console for the band, as with a larger stage area each band member will require at least one monitor wedge. The auxiliary send system of the FOH console will not be able to cope with these demands alone as it will have to deal with several effects units.

The FOH console will have a large number of mic/line inputs, plus a large number of matrix outputs so that a complex range of speaker clusters can be placed around the auditorium.

SMALL VENUES

FIG. 4.2



MEDIUM SIZED VENUES

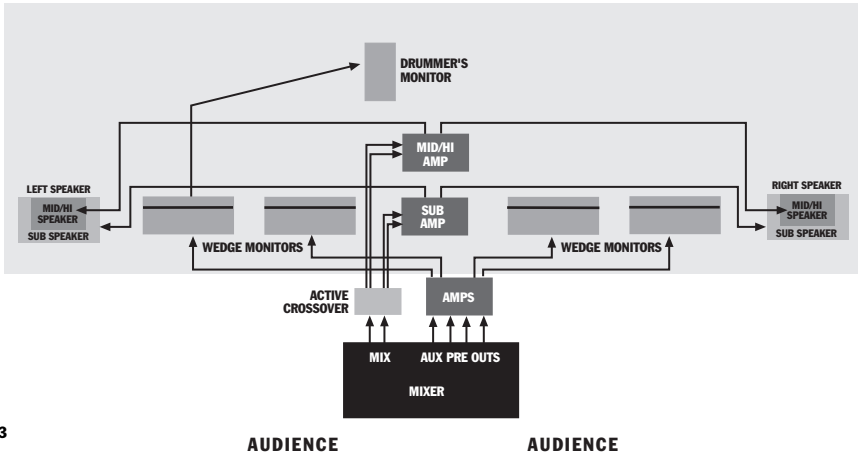


FIG. 4.3

LARGER VENUES

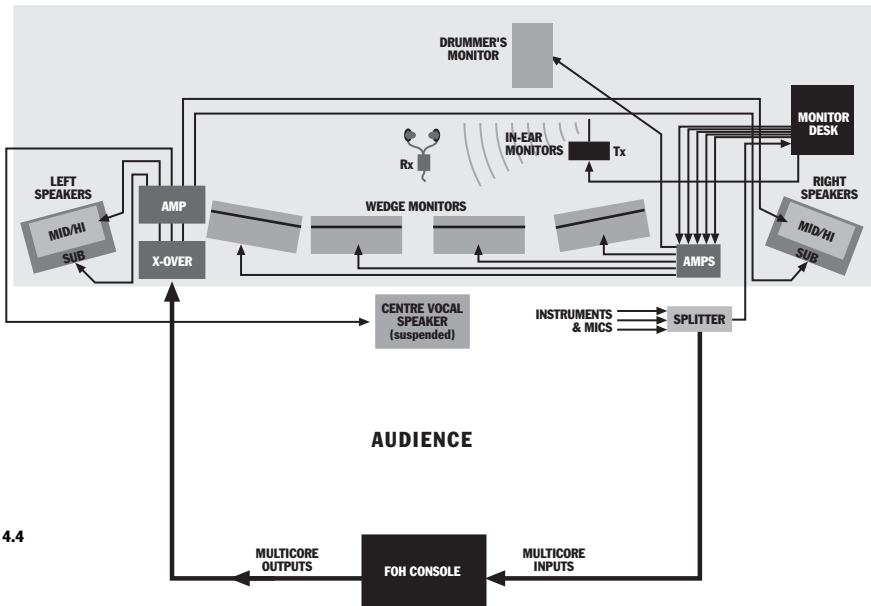


FIG. 4.4

C. Recording Live

In some situations, you may want to record a performance. Depending on the situation, the feed for recording may come from the FOH mixer, microphone splitter boxes, or your own microphones which have been set up alongside those of the band.

The diagram below shows a typical example of the sound sources being split between FOH and Recording. The recording console operates independently from the FOH mixer.

NB: When using Folio SX it will be necessary to re-patch for multitrack playback.

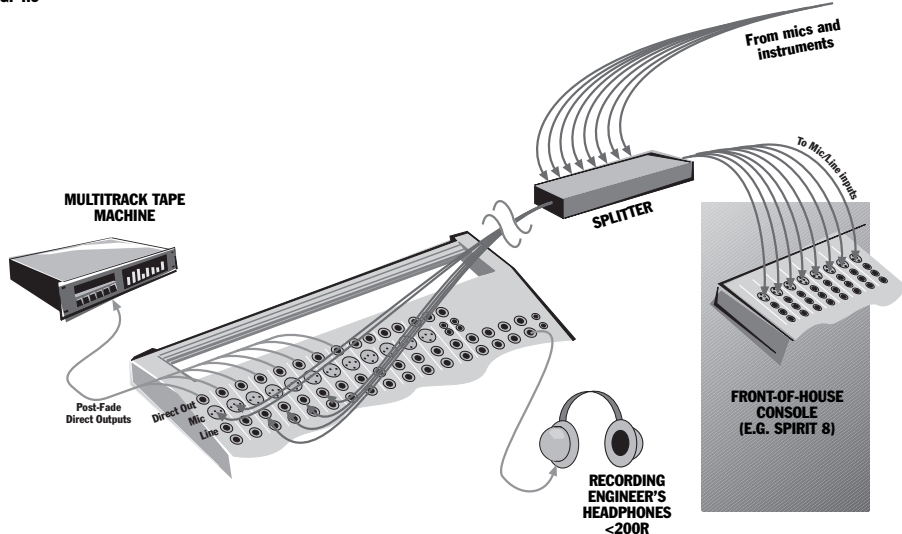
NB: Subgroups can be used for submixing many inputs (e.g. drums) to a multitrack input. This is useful when tape track availability is limited.

Hints & Tips

- Try to locate the mixer in a different room to the performance to avoid distraction from the live sound. If this is not possible, use a good pair of noise-excluding headphones for monitoring.
- Wherever possible, take feeds from mic splitters - this will provide clean, low-noise signals suitable for recording.
- Often, Tape Sends are unbalanced, so keep signal paths as short as possible between output and recorder to avoid interference.
- If there aren't enough microphones, use a stereo pair to pick up the overall sound and the rest to emphasize individual performers.
- Use a compressor/limiter to avoid overloading the digital input of the recorder.

RECORDING LIVE

FIG. 4.5



OTHER APPLICATIONS

A. Monitor Mixing

Monitors are used to allow band members to hear themselves.

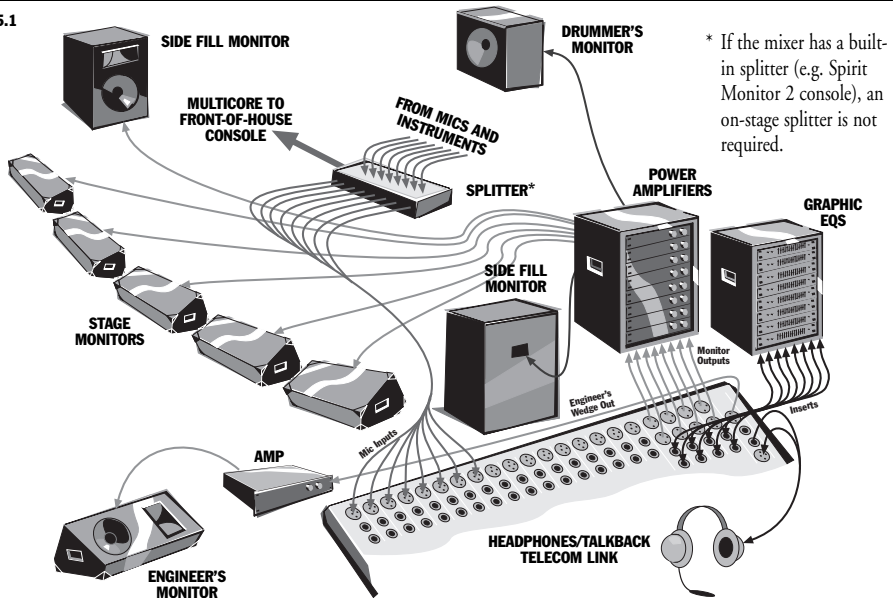
When dealing with the monitoring requirements of, say, a large live band, it is common practice to keep the monitor mix function totally separate from the Front of House console.

Some form of graphic equaliser in line with each monitor speaker is desirable as it allows troublesome frequencies to be notched out. The monitor system is rung out in exactly the same way as the main PA (see Ringing Out Section 4), and the final ringing out must be done with both the monitor and main PA systems set at their normal operating level. The monitoring console is situated off-stage and derives its feed direct from mic splitters. Note: the Spirit Monitor 2 console has its own built-in mic splitters.

- It is normal for a telecommunication link to be used between the FOH and monitor engineer so that they can talk to each other during the performance.
- Each stage monitor needs its own power amp. Keep things tidy by using rack-mounted stereo amps.
- Graphic EQs are patched via the console, like the power amps they should be rack-mounted for easy access.
- If the lead vocalist uses in-ear monitoring, he/she will be acoustically isolated, so it's a good idea to feed audience pick-up mics into his/her mix to provide a sense of involvement.
- 'Side fills' are often used where monitoring is required over a large stage area, floor space is at a premium, and too many wedge monitors would simply clutter things up both physically and acoustically. Don't compromise on these speakers - they'll have to work hard to punch sound through to the performers.
- The Monitor Engineer's wedge lets him hear the total foldback mix or selected parts thereof.
- A good Monitor Engineer, who is "invisible" to the audience, will always position himself so as to see visual signals from the performers.

MONITOR MIX

FIG. 5.1



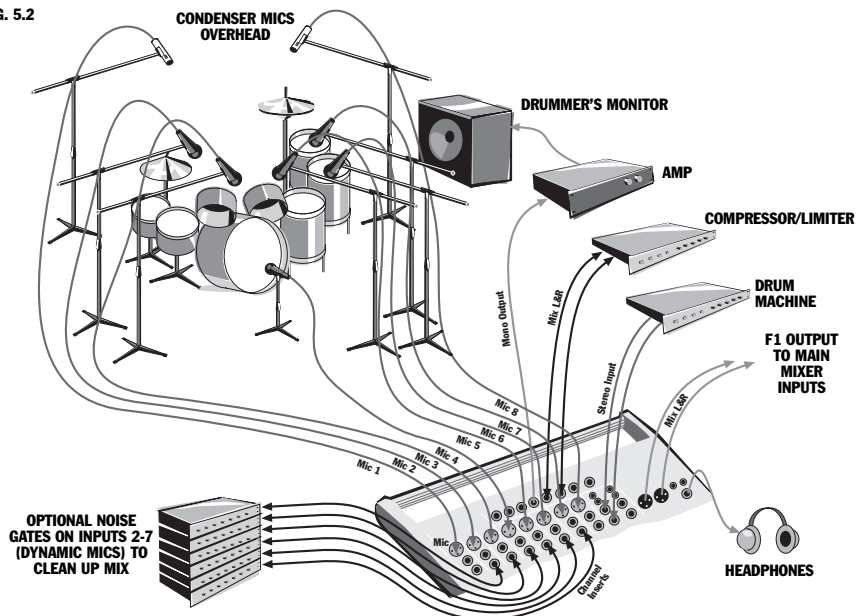
B. Submixing

There are certain groups of instruments or performers (drums, backing vocals, multi-keyboards, etc) that can be logically grouped together - to save on input channels - via a small mixer, the output of which can then be controlled by just one pair of faders on the master console.

- If a mono output is available it can be used for a drum fill or for recording purposes.
- Output from the submixer goes to the FOH console and/or may be used for a small recording set up.
- Use the Aux Returns on the FOH console to return the sub-mix. This saves valuable input channels on the FOH console.
- In the case of a drum kit where many mics are in close proximity, the use of Noise Gates will prevent spill and clean-up the mix.
- Use a Compressor/Limiter to maintain a consistent level.

SUBMIXING

FIG. 5.2



IN THE STUDIO

A. Essentials & Ergonomics

Think about room layout and equipment. No, we're not going to plan your studio for you, but here are a few pointers:

- If you play keyboards, set them up so that you can reach the mixer.
- Position your effects and synth modules within arms length.
- If you use a computer, position the screen so as to avoid reflections. Do not position speakers near the screen unless they are magnetically compensated or shielded.
- If the room is too 'live', deaden it with drapes or soft furnishings.
- For best results, use dedicated nearfield monitors.
- Don't use large speakers in a small room - they'll sound wrong at low frequencies.
- Do use a well specified power amp (minimum 50 watts per channel).
- Don't compromise on a weedy amp: it will distort at high levels and may damage the speakers.

B. Tape Machines & Recording Media

Basically, you'll need two types of tape machine: a **multitrack recorder** for recording the individual parts of the performance in readiness for mixdown onto a **2-track recorder** for mastering. There are both analogue and digital models available. The final choice must be based on individual requirements.

C. The Console

Studio work presents additional problems for a mixing console in that it has to deal with a two stage process requiring very different skills.

- 1 Recording - Sound sources have to be captured on multitrack tape. This process will include ensuring that the cleanest strongest signal is being recorded to tape, without overload and distortion, optimising the sound of the recorded signal with EQ, signal processing and effects, monitoring the recorded sources, and creating a headphone mix for the musicians to ensure the best possible performance from them.
- 2 Mixdown - All the recorded sound sources as well as any "live" media coming from sequencers, drum machines or samplers must then be blended together using EQ, level, pan and effects and mastered down to a two-track device to create a "final mix". This process bears some similarities to mixing a band - minus the audience, the live performance and poor venue acoustics!

If you have seen any T.V. shows including footage of commercial recording studios you may be forgiven for thinking that good multitrack recordings are only possible using a mammoth console. This does not have to be the case! Professional sounding results can be achieved, albeit with some repatching between recording and mixdown stages, using a relatively small multipurpose mixer. However, to achieve professional results the mixer must be equipped with either (and preferably both):

- Direct outs
- Groups/Subs

When purchasing a console for both live and recording work, ensuring these facilities are available will save you having to buy a dedicated recording console until your requirements become more sophisticated.

D. Simple Multitrack Recording

The diagram below shows a simple recording set-up using a multipurpose console equipped with direct outs and a pair of subgroups. The sound from instruments or voices is taken straight out to be recorded by the multitrack, with recorded signals being returned from the multitrack's channels into spare inputs of the mixer so they can be monitored. Alternatively, backing vocals or grouped instruments such as drumkits may be recorded to single or pairs of tracks by subgrouping them and connecting the mixer's group outputs to the multitrack device.

The engineer monitors both performances and previously recorded material through a monitor amp and speakers, with the performers getting their own separate foldback mix through the auxiliary sends.

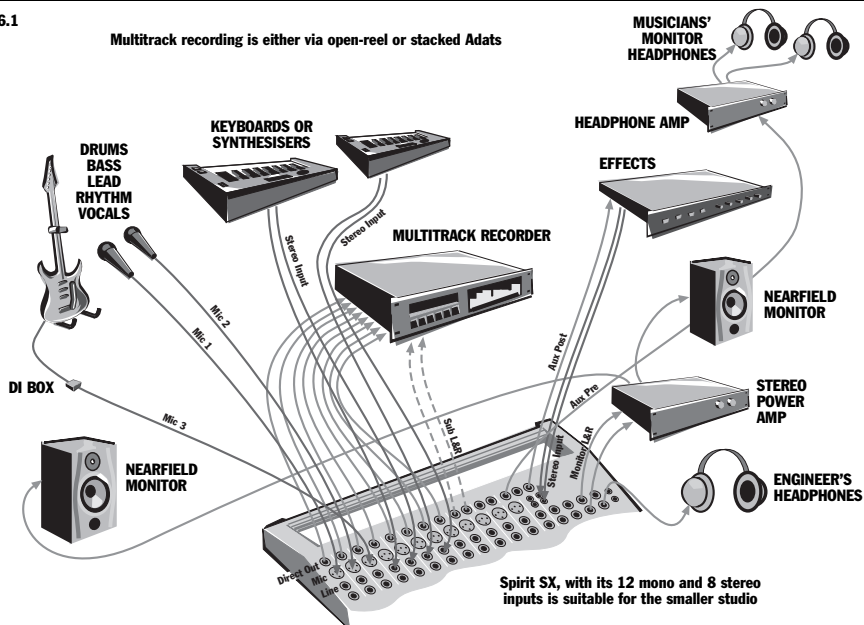
Hints and Tips when Recording:

- If you are recording as a solo performer on a budget, you can avoid the expense of buying a separate amp to create a headphone mix. Plug your headphones into the console's headphone connector and use its monitor mix for your foldback. Alter channel fader levels as you wish to achieve optimum headphone levels for your performance.
- If your console is not large enough to cope with every multitrack send and return, connect only as many Direct Outs as you need per take. For example, if you are recording solo you will only be recording one instrument at a time anyway, so a maximum of only two direct outs will be required for stereo instruments, and one for mono ones. The same channel direct outs may then be repatched to adjacent multitrack tape ins to record new tracks. This should leave enough channels free to monitor all your recorded tracks.
- If you run out of tape tracks, group instruments together. For example a fully mic'd up drumkit can be recorded in stereo to two tape tracks via a pair of groups, or if you are really stretched you could do this with the entire rhythm section, including bass and rhythm guitar. However, it is then essential to mix the balance between the instruments accurately as, once recorded, they can never be individually altered again.
- If you have only one effects unit and you need it to create a variety of different sounds, it may be necessary to record the instrument with effects included. Again, remember that once you have done this there is no going back, so wherever possible it is best to record "dry" and buy a second effects unit if you can. If you must record "wet", look at you

MULTITRACK RECORDING

FIG. 6.1

Multitrack recording is either via open-reel or stacked Adats



mixer's block diagram and use outputs coming after the effects return for this purpose.

- Do not record in the same room in which you are playing unless your monitor speakers are muted. At the very least, your recorded track will pick up the mix from the monitor speakers, but more likely howl-round and feedback will occur which will damage your equipment. If you are recording a band, it is best to put them in an entirely different room altogether.
- Setting recording levels - for the best results, as it is important to set the highest record levels you can on your multitrack without getting overload or distortion. If you set levels too low, you will end up with a weak signal and background hiss. All multitrack recorders allow you to set record levels before a take. Consult the recorder's manual as to how best to achieve this.

E. Simple Multitrack Mixdown

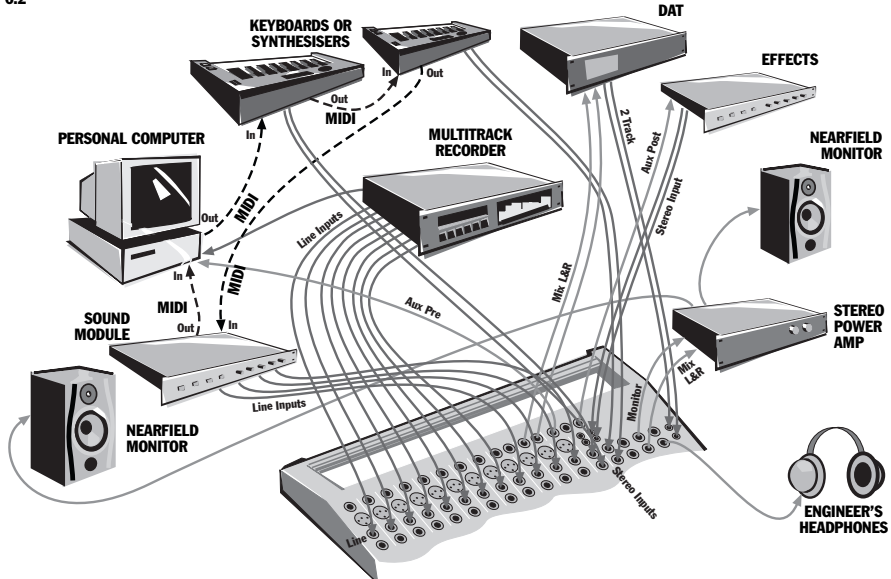
The diagram below shows how a simple set-up will look for the mixdown process. Some repatching has occurred to free up the input channels which were used as multitrack tape sends. Tape returns can then be plugged into the mixer in sequence from channel 1 upwards, leaving any spare inputs for sequenced MIDI instruments. Effects, amps and speakers may be left as before.



NB: Mixdown hints and tips may be found in "Creating a Mix" at the end of this section.

SIMPLE MULTITRACK MIXDOWN

FIG. 6.2



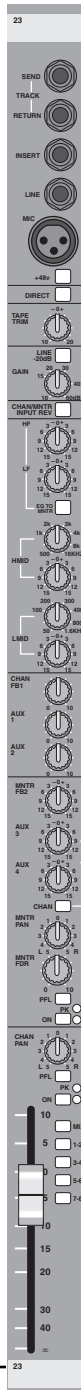
F. Using a Dedicated “In-Line” Mixing Console

For recording projects beyond 8 track, a multipurpose console is usually inadequate, being unable to cope with the additional multitrack sends and returns and with all the repatching that is required between recording and mixdown. In such cases, a dedicated “in-line” recording console is necessary. An example of the input strip of such a console is shown here.

Virtually all of the features and facilities are identical to a standard mixer - except one: as well as including full channel input facilities and a direct out (here called a tape send), the strip also includes an extra input for a multitrack tape return as well as some basic rotary level control and pan facilities for that input. This second input is known as the **Monitor Input** or **Monitor Return**. Using this technique allows a signal to and from a multitrack to be handled by one input strip, saving space and avoiding the confusion of having to find corresponding send and return signals in different areas of the console.

The major advantage of using an “in-line” recording console is that repatching is unnecessary. This is because both channel and tape return inputs can be swapped (using the switch marked “Chan/Mntr Input Rev”), giving the signal coming from multitrack all the EQ, Auxiliaries and the linear fader of the channel input for the mixdown process. This also leaves the monitor input free for sequenced MIDI gear such as keyboards. If more facilities are required for these sound sources, then EQ and auxiliaries may be shared between the two inputs.

With two inputs per channel, a 16 channel “in-line” console actually has 32 inputs available. This high input count and compactness has made “in-line” consoles extremely popular with project studios, programming and remixing suites and commercially successful bands’ home studios. With prices tumbling all the time, “in-line” consoles are now barely more expensive than standard designs.



Multitrack Recording and Mixing with an “In-Line” Console

A more complex recording set up with an “in-line” console is shown opposite in Fig 6.4. Both multitrack ins and outs are plugged into the same channel strip, avoiding the need for repatching, whilst for sound proofing purposes, musicians are recorded in a separate room. Effects and signal processors are connected in an identical way to any other console via auxiliary sends and returns and insert points.

G. Recording Instruments and Voices

VOCALS

- Use a cardioid condenser mic positioned 9 inches (225mm) from the singer.
- A pop shield will reduce explosive ‘p’ and ‘t’ sounds.
- If sibilance is a problem, change to a dynamic mic or move the singer back from the mic.

Recommended effects/processor settings:

EQ: Not normally required. But, if necessary, use the HPF (High Pass Filter) to reduce rumble.

Compressor: Attack as fast as possible; Release around 0.5S, ratio between 4:1 and 8:1.

Reverb: Try a decay time of around 3 seconds and a pre-delay of 50mS.

DRUMS

- Place mics 2 inches (50mm) from the heads of snare and kick drum.
- For the kick drum, place the mic inside - pointing directly at where the beater strikes the drumhead.
- To fully mic a kit, use separate mics on all toms and hats.
- Use condenser mics 5ft (1.5m) overhead, spaced around 5ft (1.5m) apart, to pick up the entire drum sound, cymbals and “ambience”.

Recommended effects/processor settings:

EQ: Boost at: 80Hz to add weight to kick drums, 6kHz to add sizzle to cymbals or edge to a snare. Cut at 250-300Hz to reduce boxiness on a kick drum or low toms.

Gate: Fast attack setting to allow percussive transients to pass through. Precise settings will depend on microphone type and placement.

Reverb: Keep kick drum ‘dry’. Try a percussion plate setting with a 2.5S decay time on other drums.

FIG. 6.3

ELECTRIC GUITAR

- Some players prefer the sound of a valve amplifier, so be prepared to mic up the speaker cabinet using a cardioid dynamic mic.
- Experiment with mic positioning to achieve the desired sound.
- If preferred, the guitar can be DI'd via a recording preamp which incorporates an amp simulator.

Recommended effects/processor settings:

EQ: Boost at: 120Hz to add 'thump' to rock guitars, 2-3kHz to add bite, 5-7 kHz to add zing to clean rhythm sound. Cut at: 200-300Hz to reduce boxiness, 4kHz and above to reduce buzziness.

Compressor: Attack between 10 and 50mS; Release, around 0.3S, Ratio, between 4:1 and 12:1. Because of the noise generated by a typical electric guitar, use in conjunction with a gate or expander is advised.

Reverb: Plate or room, 1.5 to 4S; 30 to 60mS pre-delay.

ACOUSTIC GUITAR

- Use the best mic that you can, preferably a condenser type.
- For a natural tone, position the mic between 12-18ins from the guitar, aiming at where the neck joins the body.
- If recording in stereo, point a second mic towards the

centre of the neck, about 12-18ins from the instrument.

- Acoustic guitars sound best in slightly live rooms, if necessary place a piece of acoustically reflective board beneath the player.

Recommended effects/processor settings:

EQ: Boost: between 5kHz and 10kHz to add sparkle. Cut between: 1kHz and 3kHz to reduce harshness, 100 and 200Hz to reduce boom. In busy pop mixes you can cut the low end to produce a more cutting rhythm sound.

Compressor: Attack 20 mS; Release, around 0.5S, Ratio, between 4:1 and 12:1.

Reverb: Bright setting such as Plate to add vitality. Decay time of between 2 to 3S.

BASS GUITAR

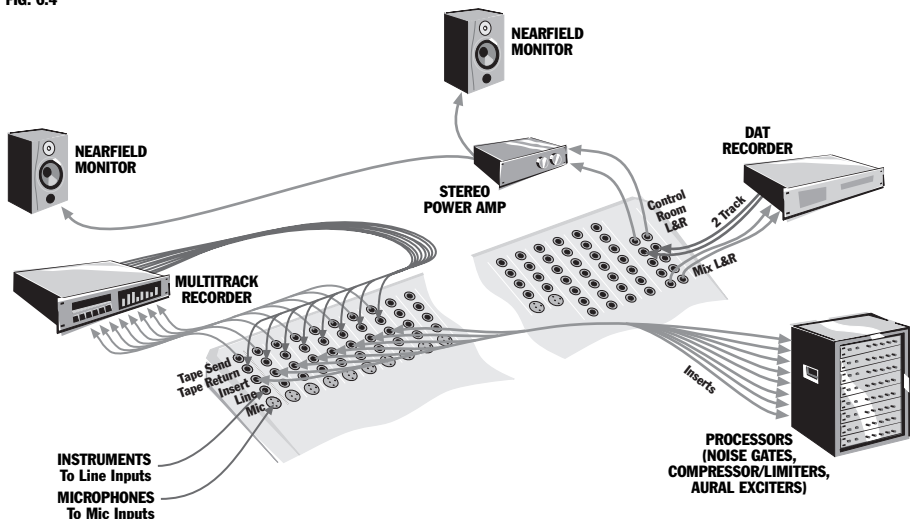
- Most engineers DI the bass via an active DI box and a compressor. This provides a clearer sound.
- Use the compressor to keep signal peaks under control.
- Check the player's technique; the harder the instrument is played, the brighter the tone.
- Consider the use of a budget graphic EQ.

Recommended effects/processor settings:

EQ: Boost: at 80-100Hz to add more weight and punch, between 2 and 4kHz to add edge. Cut: below 50Hz to

MULTITRACK RECORDING & MIXDOWN

FIG. 6.4



reduce unwanted rumble, between 180 and 250Hz to reduce boxiness.

Compressor: Attack around 50mS; Release, around 0.4S, Ratio, between 4:1 and 12:1.

KEYBOARDS

- Most electronic keyboards can be plugged directly into the line inputs of the mixing console.
- Bear in mind that the majority of contemporary synthesizers etc, have stereo outputs and will require two mixer channels.
- Most synthesizer sounds can be used without compression, though they do benefit from effects such as reverb or chorus.
- Overdriven keyboard sound may be created by feeding the signal via guitar recording preamp.

H. Planning a Session

- You have a lot to remember during a session, so create a track sheet to keep a log of what instrument is recorded onto what tape track, plus other relevant information.
- Record rhythm sections first; drums, bass, and rhythm guitar.
- Add vocals, solos, and additional instrumentation as overdubs.
- Decide whether you want to add effects at the mixing stage or while recording. If you can, try to keep a copy of the original “dry signal” on tape. You may wish to remix at a later date!
- When recording vocals, ask the singer what instruments they most need to hear in the headphone mix.

I. Creating a Mix

Go into ‘neutral’ before you start off -

- Set all the Aux Sends to zero.
- Set all EQ controls to their central positions.
- Pull all the faders down.
- All routing buttons ‘up’.

Organize your Subgroups

- Put logical groups of sounds together.
- Route drums to a stereo sub-group.
- Consider grouping backing vocals.
- Group multiple keyboards.

Metering

- Use the PFL metering system for each channel in turn to optimize the gain setting.
- The PFL should just go into yellow band of the meter

section, although peaking into the red area is acceptable.

- Check all the effects units for correct input levels.
- If fitted, use the Solo In Place function to check individual channels in isolation while retaining their original pan and level settings.

J. Balancing the Mix

If you don't have a lot of mixing experience, it can help to set up the drums and bass balance first, then move onto the vocals and the other instruments. Don't worry about fine tuning the EQ or effects until your dry mix is somewhere near right.

- Satisfy yourself that the mix is working in mono. Check for Phase problems.
- Pan bass drums, bass guitar and lead vocals to centre - this will stabilize the mix.
- Spread other instruments across the stereo stage as required, including backing singers.
- EQ the mix as required.
- Now add stereo effects as necessary to add to the illusion of space and width.
- Check the balance of your final mix by listening to it from the next room through the adjoining door: for some reason, this often shows up whether the vocals are too loud or quiet.

Hints & Tips

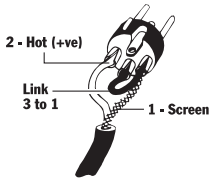
- Clean the heads of analogue tape machines before every session. Use cotton buds dipped in Isopropyl Alcohol.
- Check all instrument tunings before each take, because they have a tendency to change as the room warms up.
- Make a pop shield from stocking material stretched across a wire frame. This will minimise vocal “popping”.
- Don't skimp on cables and connectors; these can be a source of noise.

WIRING & CONNECTORS

Faulty connectors and cabling are some of the most frequent sources of noise and poor sounding systems. The following section should help you connect your system correctly. It's also worth spending a little time referring to all of your user manuals, as wiring conventions can vary between manufacturers - see diagrams.

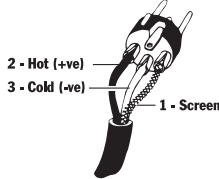
BALANCED AND UNBALANCED MIC INPUTS

FIG. 7.1



UNBALANCED INPUT

FIG. 7.2



BALANCED INPUT

Soundcraft uses XLR sockets for its balanced mic inputs. The wiring convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (-ve).

Balancing is a method of audio connection which cancels any interference in a signal, to give low noise operation. This is achieved by using a 2-conductor mic cable, usually surrounded by a shield, in which the 'hot' and 'cold' signals are opposite polarity. Any interference picked up will be of the same polarity on both hot and cold wires and will be rejected by the mic input's Difference Amplifier. You may use unbalanced sources when wired as shown. However, do not use unbalanced sources with Phantom Power switched on. The voltage on Pins 2 & 3 of the XLR connector may cause serious damage.

BALANCED AND UNBALANCED LINE INPUTS

Line inputs accept 'A' Gauge, 3-pole (Tip, Ring, Sleeve) 1/4 inch jack wired as shown in Fig. 7.3.

FIG. 7.3

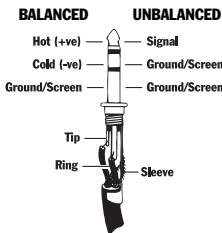
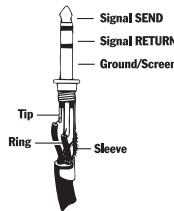


FIG. 7.4



INSERTS

A Mixer insert point is a single, 'A' Gauge, 3-pole (stereo), switched jack socket (not unlike the headphone socket on a hi-fi amplifier). When a 3-pole jack is inserted the signal path is interrupted. The signal is taken out of the mixer via the plug tip, through an external piece of equipment and then back to the mixer on the ring of the plug. A special Y-cord is required which has the stereo jack at one end and two mono jacks, for the processor's input and output, at the other. See Fig. 7.4.

GROUND COMPENSATED OUTPUTS

Ground compensated outputs may, to all intents and purposes, be treated as balanced outputs. Ground compensation helps avoid hum loops when the console is feeding into an unbalanced piece of equipment. Essentially, the Ground Compensated output has three connections, much like a conventional balanced output, except that the pin normally designated 'cold' acts as a 'ground sense' line enabling it to sense and cancel any ground hum present at the output.

BALANCED CONNECTION

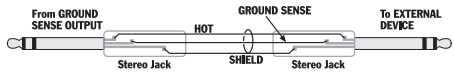


FIG. 7.5

UNBALANCED CONNECTION

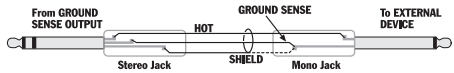


FIG. 7.6

The convention for XLRs is: Pin 1 - Shield, Pin 2 - Hot, Pin 3 - Ground Sense. For jacks, the wiring convention is: Tip - Hot, Ring - Ground Sense, Sleeve - Shield.

For use with balanced destinations, the Ground Sense output may be treated as 'cold' allowing the connection to be made normally. Where the destination has an unbalanced jack input, a two-core (balanced-type) lead should be made up as shown. Unbalanced jacks may also be plugged directly into Ground Compensated Output jack sockets, but the benefit of hum rejection will be lost.

IMPEDANCE BALANCED OUTPUTS

Impedance Balanced Outputs are configured as normal balanced outputs: Pin 1 - Shield, Pin 2 - Hot (+ve) and Pin 3 - Cold (-ve). See Fig. 7.2.

Impedance Balanced Outputs work on the principle that hot and cold terminals have the same resistance. When impedance balanced outputs are used with a balanced input, good rejection is achieved for both common-mode ground voltages and electrostatic interference.

Note that for unbalanced operation the screen of the cable is wired to both the Ring and the Sleeve of the jack.

Note: The cold terminal can be either shorted to ground locally or left open-circuit for balanced and unbalanced operation.

GLOSSARY

ACOUSTIC FEEDBACK (HOWLROUND)

A whistling or howling noise caused by an amplified signal 'feeding back' into the amplification chain via a microphone or guitar pick-up.

ACTIVE DI BOX

A device which permits Direct Injection of signals from guitars, etc, into the console. Incorporates circuitry to adjust gain and provide impedance matching. Requires power and may be battery driven or sometimes 'phantom powered' from a console.

AFL (AFTER FADE LISTEN)

A function that allows the operator to monitor a post-fade signal. Used with Aux Masters.

AMPLIFIER

Device that increases the level of an electrical signal.

AMPLITUDE

Signal level, usually in volts.

ANALOGUE

Analogy (n.): correspondence or partial similarity, using physical variables. For example; an analogue tape recorder stores sound on tape in the form of a magnetic pattern which is a replica of the original musical waveform.

ASSIGN

On a mixing console, to switch or route a signal to a particular signal path or combination of signal paths.

ATTENUATE

To decrease the level of a signal.

AUXILIARY SEND

Level control feeding a dedicated bus for driving external effects or a foldback monitoring system. An output from the console comprising a mix of signals from channels derived independently of the main stereo/group mixes. Typically the feeds to the mix are implemented on rotary level controls.

BACK-LINE

Stage parlance for the row of instrument amplifiers and loudspeaker cabinets behind the performers, e.g. guitar amps.

BALANCE

Relative level of the left and right channels of a stereo signal.

BALANCED

A method of audio connection which 'balances' the signal between two wires and a screen which carries no signal. Any interference is picked up equally by the two wires, through common mode rejection at the destination differential balanced input resulting in cancellation of the interference signal. For balancing to be effective, both the sending and receiving device must have balanced output and input stages respectively.

BANDWIDTH

A means of specifying the range of frequencies passed by an electronic device such as an amplifier, mixer or filter.

BARGRAPH

A row of LEDs calibrated to indicate signal level.

BOOST/CUT CONTROL

A single EQ control which allows the range of frequencies passing through its filter to be either amplified or attenuated. The centre position is usually the 'flat' or 'neutral' position.

BUS or BUSS

A defined set of conductors along which signals may travel. A mixer has several busses carrying the stereo mix, the groups, the PFL signal, the aux sends, etc.

CAPACITOR

See Condenser

CARDIOID PATTERN

The 'heart-shaped' polar response of a microphone meaning that most of the sound is picked up from the front. Mainly used for stage vocals or in any situation where sound has to be picked up from a concentrated area, i.e. drums.

CHANNEL

A strip of controls in a mixing console relating to a single mono input or a stereo input.

CHIP

Integrated circuit; a multi-pinned device consisting of many circuits encapsulated in plastic.

CHORUS

Effect created by doubling a signal and adding delay and pitch modulation.

CLIPPING

Severe form of audio distortion which is the result of signal peaks exceeding the amplifier capacity. Normally caused by a limitation of the unit's power supply.

CLONE

Exact duplicate. Often refers to digital copies of digital tapes.

CONDENSER

Electrical component exhibiting capacitance (the ability to temporarily store electric current) and block direct current.

CONDENSER MICROPHONE

A type of microphone which picks up sound via a thin, flexible diaphragm placed in proximity to a metal plate - as opposed to the rigid diaphragm-and-coil system used by dynamic microphones. Condenser mics are very sensitive, especially to distant sounds and high frequencies. They have to be powered, which can be achieved by batteries, but for professional use a 48v DC PHANTOM POWER supply is provided from the console via the balanced mic cable.

CONDUCTOR

A thing that conducts or transmits heat or electricity.

COMPRESSOR

A device designed to control or reduce the dynamic range of an audio signal.

CROSSOVER

A passive circuit, normally built into a speaker system which divides the full-range audio signal from an amplifier in order to feed the individual drive units, ie: bass, midrange and treble.

CUEING

To put a piece of equipment in readiness to play a particular part of the recording material. Assisted on a mixing console by use of the PFL (Pre-Fade Listen) facility.

CUT-OFF FREQUENCY

The frequency at which the gain of an amplifier or filter has fallen by 3dB.

DAT (DIGITAL AUDIO TAPE)

High quality cassette based recording format which stores signals digitally and therefore provides very high quality sound. Originally touted for consumer use, but now firmly ensconced as a professional tool.

dB (DECIBEL)

A ratio of two signal levels. Can be in Voltage, Watts or Current units.

dBm

Variation on dB referenced to 0dB = 1mW into 600 ohms.

dBu

Variation on dB referenced to 0dB = 0.775 volts.

dBV

Variation on dB referenced to 0dB = 1 Volt.

DETENT

In audio terms a click-stop in the travel of a rotary or slide control, normally used to indicate 'centre stereo' on pan-pots or 'zero boost/cut' on EQ controls.

DI BOX

A device allowing connections as explained below.

DI (DIRECT INJECTION)

The practice of connecting an electric musical instrument directly to the input of the mixing console, rather than to an amplifier and loudspeaker which is covered by a microphone feeding the console.

DIGITAL DELAY

The creation of delay and echo effects in the digital domain. The premise being that, as digital signals are resistant to corruption, the process will not introduce additional noise or distortion.

DIGITAL REVERB

Reverberation effects created as above.

DIGITAL

The processing and storage of signals with sound- information represented in a series of '1s' and '0s', or binary digits.

DIRECT OUTPUT

A pre-/post-fade, post-EQ line level output from the input channel, bypassing the summing amplifiers, typically for sending to individual tape tracks during recording.

DRY

Slang term for an original audio signal that has had no added effects.

DYNAMIC RANGE

The ratio in decibels between the quietest and loudest sounds in the audible range that the audio equipment will reproduce.

DYNAMIC MICROPHONE

A type which uses a thin diaphragm attached to a coil of wire arranged about a permanent magnet. Any variation in air pressure on the diaphragm will cause the coil to generate a minute electric current which then requires amplification.

EARTH

See GROUND.

EFFECTS

The use of devices to alter or process the sound to add special effects eg: reverb, normally as a mix of the original ('dry') sound and the treated ('wet') version.

EFFECTS RETURN

Additional mixer input designed to accommodate the output from an effects unit.

EFFECTS LOOP

Connection system that allows an external signal processor to be connected into the audio chain.

EFFECTS SEND

A post-fade auxiliary output used to add effects to a mix.

ELECTRET MICROPHONE

Type of condenser microphone using a permanently charged capsule.

ELECTRONIC CROSSOVER

An active device which divides the full range audio signal into several narrower frequency bands (eg: low, mid and high), which are then amplified and fed to the appropriate speaker drive units.

ENCLOSED HEADPHONES

Types that completely enclose the ears and therefore provide good exclusion of outside noise. Of particular use when monitor mixing or recording live on stage.

EQ

Abbreviation for equaliser or equalisation.

EQUALISER

A device that allows the boosting or cutting of selected bands of frequencies in the signal path.

EXPANDER

The opposite of a compressor, an expander increases the dynamic range of signals falling below a pre-determined threshold.

FADER

A linear control providing level adjustment. Favoured by professionals due to smoothness of activation and the ability to give an instant visual indication of status.

FILTER

A filter is a device or network for separating waves on the basis of their frequencies.

FOH

An acronym for Front Of House. In the entertainment world "House" is a collective term for the audience at a theatre, cinema, etc. Hence an FOH console will be situated "audience-side" of the stage. A "house" PA system refers to the main audio system responsible for the principal sound in the venue.

FOLDBACK

A feed sent back to the artistes via loudspeakers or headphones to enable them to monitor the sounds they are producing.

FOLDBACK SEND

A pre-fade auxiliary output used to set up an independent monitor mix for the performers.

FREQUENCY RESPONSE

The variation in gain of a device with frequency.

FSK (Frequency Shift Keying)

A method of synchronisation which generates a series of electronic tones related to the tempo of the music. These tones may then be record on a spare track of the multitrack recorder.

FX UNIT

Slang term for Effects Unit. Typical effects units are delays, reverbs, pitch shifters, and chorus units.

GAIN

Gain is the factor of how much the level of a signal is increased or amplified. Normally expressed in decibels.

GATE

A user-adjustable electronic device that switches off the signal path when the signal falls below a certain predetermined level or threshold.

Typically used to ensure silence between pauses in the signal during vocal passages or to prevent 'spill' between the close-proximity, multiple mics on a drum kit.

GRAPHIC EQUALISER

Device incorporating multiple narrow-band circuits allowing boost and cut of predetermined frequencies. Vertical fader controls are used which provide a 'graphic' representation of the adjustments across the frequency range.

GROUND COMPENSATION

A technique used to cancel out the effect of ground loops caused by connections to external equipment.

GROUND

Ground and Earth are often assumed to be the same thing, but they are not. Earth is for electrical safety, while Ground is the point of zero voltage in a circuit or system.

GROUND LOOP

A ground loop occurs when there are too many ground points, allowing small electrical currents to flow.

GROUP

An output into which a group of signals can be mixed.

HEADROOM

The available signal range above the nominal level before clipping occurs.

HERTZ (Hz)

Cycles (or vibrations) per second.

HIGH PASS FILTER

A filter that rejects low frequencies below a set frequency, typically 100Hz.

IMPEDANCE

The AC resistance of a circuit which has both resistive and reactive components.

IMPEDANCE BALANCING

A technique used to minimise the effect of hum and interference when connecting to external balanced inputs.

INDUCTOR

Reactive component that presents an increasing impedance with frequency. A coil in a loudspeaker crossover is an inductor.

INSERT POINT

A break point in the signal path to allow the connection of external devices, for example signal processors or to another mixer.

K OHM, K Ω or kHz

x 1000 ohms, x 1000 ohms and x 1000Hz respectively.

LINE LEVEL

Signals at a nominal level of -10dBV to +4dBu, usually coming from a low impedance source such as keyboards, drum machines, etc.

mA (milliampere)

One thousandth of an ampere, a measure for small electrical currents.

MIC SPLITTER

A device which divides the output from a microphone in order to supply two signals, for example; FOH console and recording mixer or monitor console.

MIDBAND

The range of frequencies to which the human ear is most sensitive.

MIDI

Musical Instrument Digital Interface.

MIXDOWN

The process of taking the outputs from a multitrack recorder, processing as required and combining all elements to create a stereo 'master'.

MONITOR LOUDSPEAKER

Any high quality loudspeaker which is used to check the quality or status of the signal.

MTC (MIDI Time Code)

An interpretation of SMPTE allowing the time code to come in as part of the MIDI data stream.

MULTICORE

A cable with multiple cores allowing signals to be carried independently but within the same physical outer casing.

MUTE GROUPS

A method of combining the on/off status of a selection of channels under a single control button.

NEARFIELD MONITOR

A high quality, compact loudspeaker designed for use at a distance of three to four feet from the operator. Their use ensures that detrimental room effects are minimised.

NORMALISE

A socket is said to be normalised when it is wired in such a way that the original signal path is maintained unless a plug is inserted into the socket. The most common examples of normalised connectors are the INSERT POINTS found on mixing consoles.

OSCILLATOR

A tone generator for test and line-up purposes.

OVERDUB

To add another part to a multitrack recording or replace one of the existing parts.

OVERLOAD

To exceed the operating capacity of an electronic or electrical circuit.

PAN (POT)

Abbreviation of 'panorama': controls levels sent to left and right outputs. Allows positioning of signals within the stereo sound stage.

PARAMETRIC EQUALISER

A graphic equaliser in which the cut/boost, frequency and bandwidth are all adjustable.

PASSIVE

A circuit or component which does not amplify the signal or is not powered.

PATCH BAY

A system of panel mounted connectors used to bring inputs and outputs to a central point from where they can be routed using plug-in patch cords.

PATCH CORD

Short cable used with patch bays.

PEAKING

A signal of the maximum displacement from its mean (average) position.

PHANTOM POWER

The +48v DC voltage applied equally to the two signal pins of a balanced mic input to provide powering for condenser microphones.

PHASE

Phase is the fraction of the whole period that has elapsed, measured from a fixed datum. A term used to describe the relationship of two audio signals: in-phase signals reinforce each other, out-of-phase signals result in cancellation.

PHONO PLUG

A hi-fi connector developed by RCA and used extensively on semi-pro recording equipment.

POLARITY

The orientation of the positive and negative poles of an audio connection. Normally, connections are made positive to positive, negative to negative and this would ensure correct polarity. If this is reversed the result will be out-of-phase signals (see PHASE above).

POP SHIELD

A device used in the studio, consisting of a thin mesh placed between the microphone and vocalist in order to reduce the 'explosive' effects of 'P' and 'T' sound

POST-FADE

The point in the signal path after the channel or master fader and therefore affected by fader position.

PRE-FADE LISTEN (PFL)

A function that allows the operator to monitor the pre-fade signal in a channel before it reaches the main mix.

PRE-FADE

The point in the signal path before the monitor or master position and therefore unaffected by the fader setting.

PROCESSOR

A device which affects the whole of the signal passing through it, e.g. gate, compressor or equaliser.

Q (Bandwidth)

A measure of the sharpness of a bandpass filter. The higher the value of Q, the narrower the band of frequencies that passes through the filter.

RESISTANCE

Opposition to the flow of electrical current.

REVERB

Acoustic ambience created by multiple reflections in a confined space. A diffuse, continuously smooth decay of sound.

RINGING OUT

The process of finding the problem frequencies in a room by steadily increasing the gain of the system until feedback occurs. A GRAPHIC EQUALISER is then used to reduce the offending frequencies.

ROLL-OFF

A fall in gain at the extremes of the frequency response. The rate at which a filter attenuates a signal once it has passed the filter cut-off point.

SEQUENCER

Computer-based system for the recording, editing and replay of MIDI music compositions.

SHELVING

An equaliser response affecting all frequencies above or below the break frequency i.e. a high-pass or low-pass derived response.

SHORT CIRCUIT

The situation where two electrical conductors touch.

SIBILANCE

n. sounding with a hiss. When certain phonics are exaggerated, i.e. s, sh.

SIGNAL

Electrical representation of input such as sound.

SIGNAL CHAIN

The route taken by a signal from the input to a system through to its output.

SIGNAL-TO-NOISE RATIO

An expression of the difference in level between the audio signal and the background noise of the device or system. Normally expressed in decibels.

SMPTE (Society of Motion Picture and Television Engineers)

Time code developed for the film industry but now extensively used in music and recording. SMPTE is a real-time code and is related to hours, minutes, seconds and film or video frames rather than to musical tempo.

SOLO-IN-PLACE

A function that allows the operator to listen to a selected channel on it's own but complete with all relevant effects, by automatically muting all other inputs.

SOUNDCRAFT

The name found on some of the best-value professional audio equipment around.

SOUND REINFORCEMENT

The process of amplifying or reinforcing on-stage sound (whether from already-amplified or acoustic instruments/voices) without overpowering the original sound. Suitable for smaller venues and often used solely to raise the level of the vocals above the back line and drums.

SPL (Sound Pressure Level)

Intensity of sound measured in decibels.

STEREO

Two channel system feeding left and right speakers to create the illusion of a continuous sound field. Stereo: from the Greek word for 'solid'.

STEREO RETURN

An input designed to receive any stereo line level source such as the output of effects or other external processing devices.

STRIPE

To record time code onto one track of a multitrack tape machine.

SWEEP EQ

An equaliser section (e.g. Midband EQ) which boosts or cuts a variable rather than fixed frequency.

TALKBACK

A system allowing the operator to speak to the artistes or to tape via the auxiliary or group outputs.

TAPE RETURN

A line level input provided specifically to receive the playback output of a tape machine.

TRANSIENT

An instantaneous rise in the signal level e.g. a cymbal crash or similar.

TRIM CONTROL

A variable control which gives adjustment of signal level over a limited and predetermined range usually for calibration purposes.

TRS JACKS

A 3-pole jack with Tip, Ring and Sleeve connection. Sometimes referred to as a stereo or A-gauge jack plug.

UNBALANCED

A method of audio connection which uses a single signal wire and the cable screen as the signal return. This method does not provide the same degree of noise immunity as a BALANCED connection.

WET

Slang term for a signal with added effects such as REVERBERATION, ECHO, DELAY or CHORUS.

Y-LEAD

A lead split so that one source can feed two destinations. Y-leads may also be used in console insert points in which case a stereo jack plug at one end of the lead is split into two mono jacks at the other.

2-TRACK RETURN

A line level stereo input on a mixing console designed to accept the output from a 2-Track recording device. May also be used as an additional effects return, depending on the internal routing of the mixer.

A Typical Block Diagram (Spirit SX shown)

