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 -

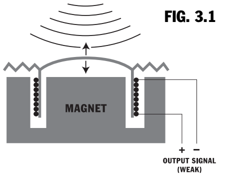


FIG. 3.1

- 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.

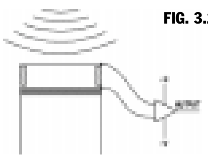


FIG. 3.2

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.

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.


FIG. 3.3

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.

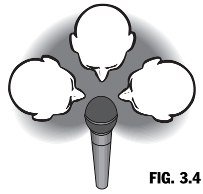


FIG. 3.4

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.

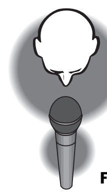


FIG. 3.5

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.

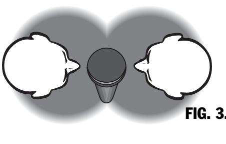


FIG. 3.6

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 preserve the best possible sound

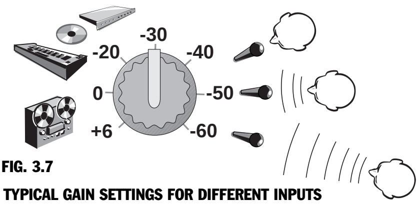


FIG. 3.7

TYPICAL GAIN SETTINGS FOR DIFFERENT INPUTS

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 Spirit input faders.
- Release PFL/Solo.
- Repeat for all other inputs.

NB: EQ affects gains 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.

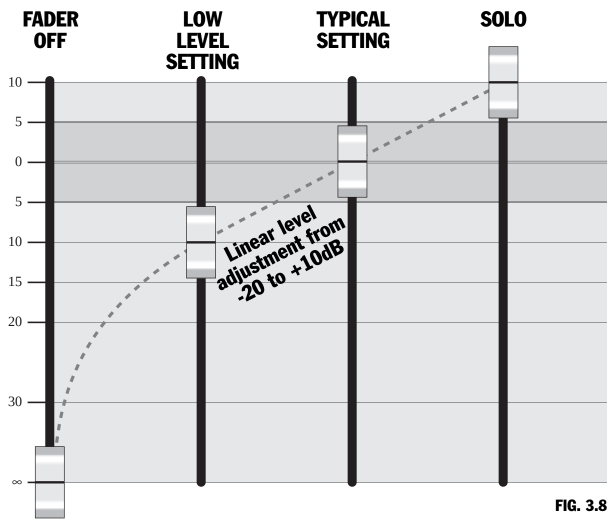


FIG. 3.8

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.

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*

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

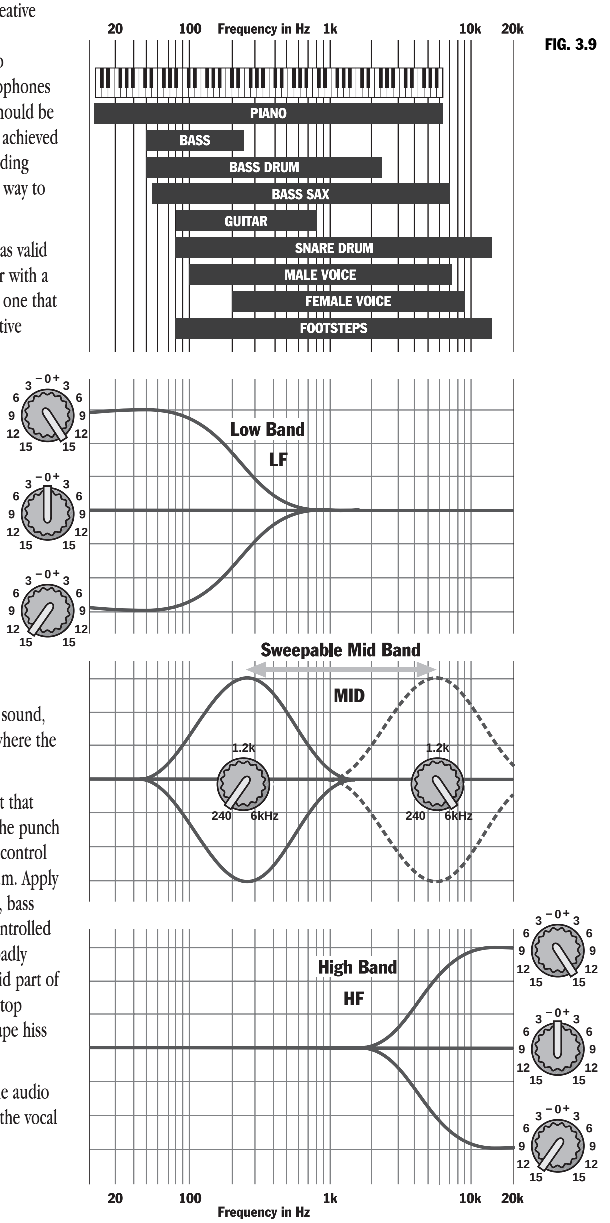


FIG. 3.9

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:

- 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.

> *Caution: when adjusting EQ, there is a danger of feedback which can cause damage to your speakers.*

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.

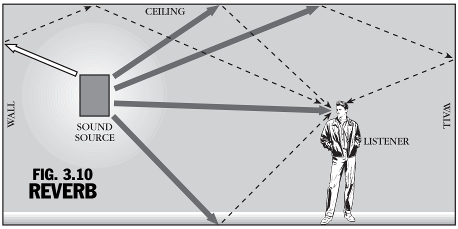


FIG. 3.10 REVERB

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

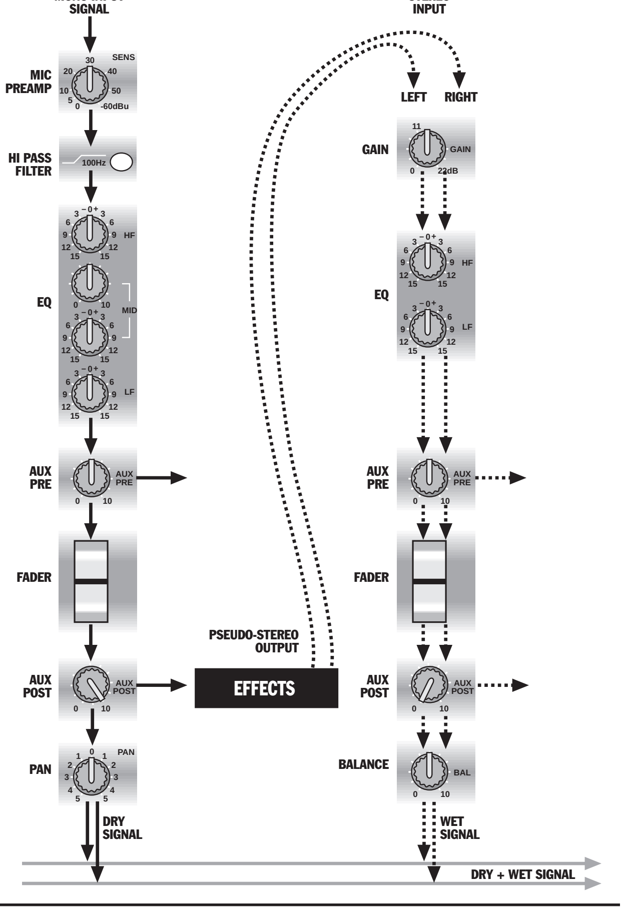
Pre-fade auxiliaries are independent of the fader so that the amount of effect will not change with new fader levels. This means you will still hear the effect even when the fader is at the bottom of its travel.

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.

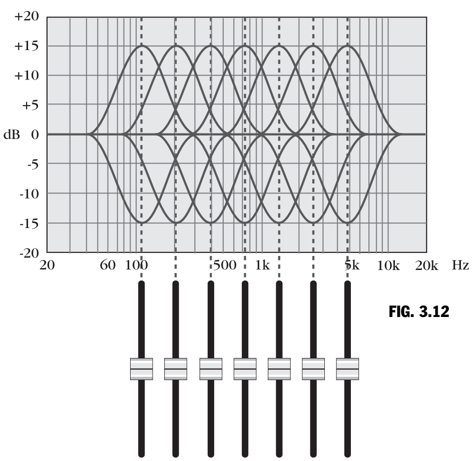


FIG. 3.12

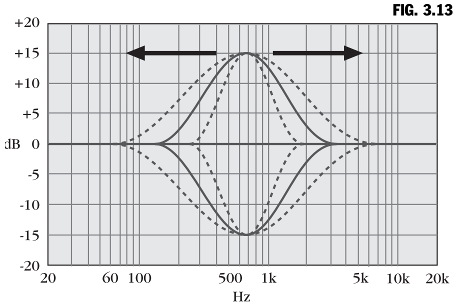
Graphic Equalisers are most often used to process the mix at live venues by notching out troublesome frequencies that may be causing feedback. They may also be used to enhance a mix at a poor sounding venue. In recording they are used to create "flat" listening environments.

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.

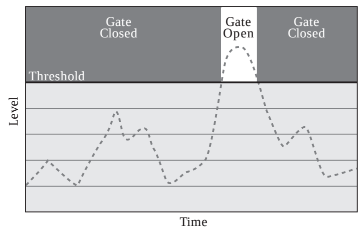


FIG. 3.14

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.

Setting up a Signal Processor

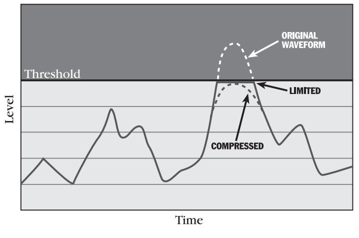


FIG. 3.15

- Connect your processor to the relevant mixer insert jack (mono, group or mix insert), using a 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.