Australian Technical Production Services

Mixing Part 2

Why part 2? And where the heck is part 1?

Part 1 originally covered setting up a PA system and plugging everything in. Unfortunately part one was on the hard drive of a laptop that was stolen and I had no back up other than a print out.

Going over the printed copy I found a number of errors making me reluctant to simply scan the document and it seemed pointless writing part 1 as a separate document so I am adding the topics covered to these notes.

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Any future updates and other notes may be found at : www.atps.net.

While you are under no obligation to send me any suggestions/updates/improvements or corrections, I would appreciate them (contact via tech@atps.net).

Of course anyone who feels like proof reading please feel free, I know my writing leaves plenty of scope for corrections in Grammar and Punctuation and of course you do get to put your name in the Credits.

As these notes are constantly under constant revision (at this stage) may I suggest that any corrections are highlighted in Red and the document sent back to me so I can apply those corrections to the latest revision.

Revision history:

04/11/2013	Corrections
21/02/2012	Corrections
04/09/2010	Balanced and unbalanced
12/07/2010	Microphones
08/07/2010	dodgy speaker connectors, Smoking cables

Credits:

This Article contains contributions by:
Richard Freeman

Drawings created using **Vector Engineer Pro-tools**
Some notes shared with other documents, such as “how to kill a PA”, “Soundcheck” and “Acoustic space”.

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Introduction

Live sound involves a series of compromises between the ideal world (where you can place speakers where you would like, the Venue is acoustically perfect, there is no spill off stage) and the reality where time and budget is limited, speakers cannot simply be placed where you want, cable runs need to be taken into consideration so they don't trip members of the public and the mix position often ends up in a corner of the room where you cannot quite hear what is going on.

In the world of live sound it is a rare gig where you can think afterwards 'everything went perfectly'.

In a studio you have a controlled acoustic environment without excessive echo but most of all you have the luxury of time (that mix didn't quite go as smoothly as I hoped, go back to the beginning of the track and lets do it again, the guitarist missed a note? No worries go back and do it again).

In live sound your first battle is often with the acoustics of the venue and if you miss a cue (didn't quite feature the Guitar at the beginning of the song? lead Vocals not quite up enough?) that's it, you've missed your chance, that moment in time is never coming back again.

So what am I saying here? Give up? Don't bother trying?

No rather quite the opposite, what I am saying is that on a good day things will be hard enough, often for reasons completely out of your control. You will miss a cue, be distracted by someone asking where the toilets are and while this is not ideal it is normal. Fortunately though with live sound the moment is gone and in many cases no-one except you remembers....

You need to have everything as good as you can from the start, so your only battle is with the expected adversaries (typically, the venue and Sound off stage), you can stay focussed and get the mix right and this means you cannot afford to be also battling a dodgy system, lack of sleep or excessive alcohol consumption as well.

Above all though, learn to live with your limitations and work with the compromises forced upon you as best you can.

As an example of the compromises I often face. I work with a Jazz/funk band playing at parties and weddings. Now in a perfect world in a perfect venue I would set my desk up maybe 2/3rds the way up the venue out where the audience are, but do you think this is an option at a wedding where appearance is paramount?

The compromise here is setting up where I can (not where I want to) and sometimes I do not even get the luxury of running a multicore out, but instead have to walk the room, listen and go back and tweak the mix (now if only I could get a digital console with Blue-tooth control and an app for that...).

Another very common compromise is the reverberant venue, where you have to keep the mix simple, do the best you can and accept that you are unlikely to achieve technical perfection (see my notes on Acoustic space where I discuss this issue further).

Notes and conventions:

For the sake of simplicity in these notes I include singers under the all encompassing term 'musicians' and so do not differentiate between playing an instrument and singing. So for example if I say 'ask the musicians to start playing' – this would include the singers as well as other instruments.

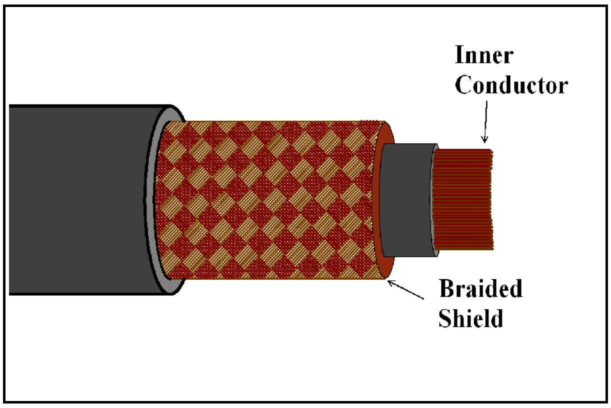
Cables

Balanced, unbalanced or downright strange?

OK maybe not the strange bit – I just threw that in to make the title sound more interesting but balanced and unbalanced?

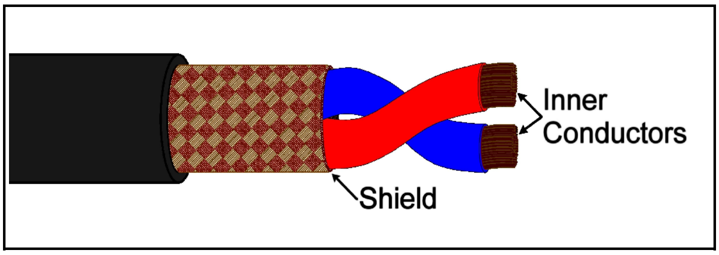
Unbalanced

This is the easiest to understand, the cable simply carries signal and ground the ground is usually attached to or is a shield which surrounds the signal wire (a Caution here: cheaper cables often have minimal shielding with some of the cheapest cables having no shielding at all and the only way you can tell this would be by stripping the cable).



As the name suggests the shield is there to shield the wire from interference.
Unfortunately there are limitations to how effective the shield is and the lower the frequency interference you wish to shield against, the thicker the shield needs to be, The trade-off is though, that the lower the frequency interference is the less effectively shorter cables pick it up, however at low signal levels (e.g microphone level) it does not take much interference to be audible.
This limits the use of unbalanced cabling to higher level signals (such as line level) and/or shorter runs (generally runs over 3 meters should be avoided).

Balanced (Microphone)



Balanced cabling consists of two signal wires – one commonly called 'in phase' and the other 'out of phase' (these names can be a tad misleading and I only use them here as this reflects common usage) surrounded by a shield. as before the shield is there to stop the signal wires picking up electrical interference but why are there two signal wires?

I could go on about in-phase and inverted signals etc. but I think the easiest way to understand how balanced cabling and circuitry 'rejects' interference is like this:

Think of the 'in phase' circuit as carrying the (wanted) signal and 'out of phase' (for the sake of this explanation) has no signal*.

Since both conductors are twisted together, any interference or noise picked up by the in phase circuit will also be picked up equally*[1] by the out of phase circuit. So if the receiving electronics takes the difference between the two signals (i.e. by subtracting the out of phase signal from the in phase signal) as the interference is common to both circuits this effectively cancels it out.

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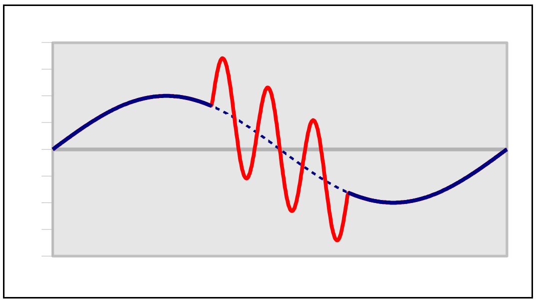
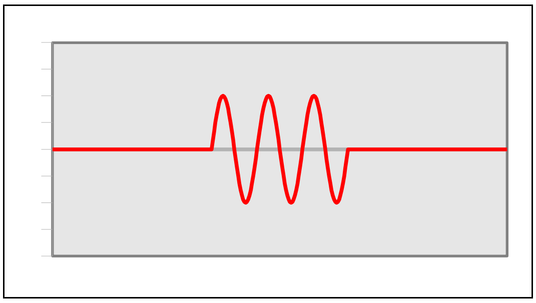


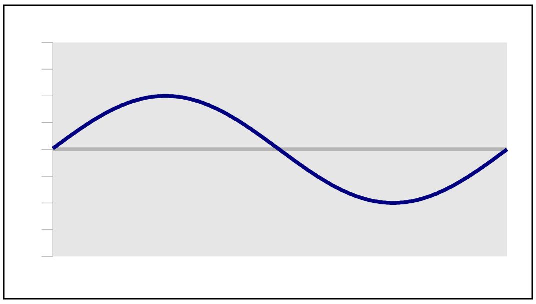
In phase (Signal + interference)

Minus



Out of phase (Interference)

Equals:



Signal only....

one thing to note here is that the ability of an input circuit to cancel out interference common to both legs is known as CMRR or Common Mode Rejection Ratio. Electronic input stages (such as the microphone pre-amp on a mixing desk) tend to have a CMRR that decreases as frequency increases, this is why PA systems can be prone to picking up higher frequency harmonics such as dimmer buzz.

*While not strictly necessary, many balanced outputs do have an equal but opposite signal on the out of phase pin, as this is not needed in order for balanced cabling to work and only potentially complicates this explanation, I have not shown that here.

*1 for this to work the electrical characteristics (impedance, reactance etc.) of both the in phase and out of phase circuits need to be exactly the same, or electrically balanced – hence the name and I while I could go into further detail this would quickly get out of the scope of this article, see “how to smash a sound system” for more details.

Phantom power

In order to power to devices such as active DI boxes and Condenser Microphones a technique called phantom powering was developed that superimposed power on the signal wires of balanced cables.

This provides power down both signal wires with current return via the ground wire, power and signal are separated using a filter (usually just a capacitor) or transformer winding.

While most balanced microphones do not mind having phantom power applied whether they need it or not, unbalanced equipment can often be damaged by phantom power and it is fair to say that balanced to unbalanced adaptor cables and phantom power are never a good combination.

Multicore cables or 'snakes'

A multicore is simply a bundle of microphone cables in a common sheath. A good multicore will have each channel (Microphone cable) insulated from the others and often an overall shield around the whole bundle. Since it contains a number of channels a multicore cable is usually big and heavy, because the microphone channels are made thinner to reduce the size and weight of the multicore this means that they are relatively delicate and need care when handling or coiling them.

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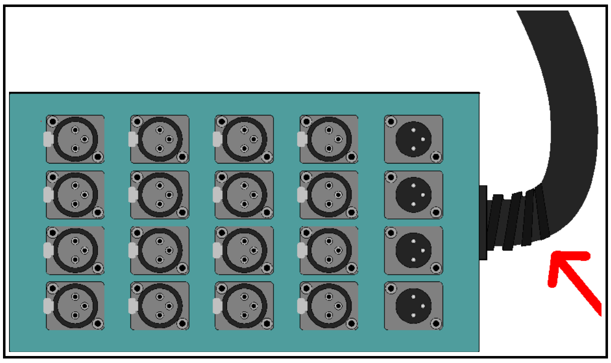


Multicore cables can be prone to failure where the cable enters the stage box. This is because the cable can be pulled into a tighter bend where it leaves the stage-box and this places more strain on the delicate wires within, as shown in the picture on the left.

Speaker Cables

Over the years a number of different connectors have been used for speaker cables –

3 pin XLR connectors

No real standard ever quite developed when using XLRs for speaker leads – the most common variation I saw used pin 1 and pin 3,

but the problem was if you plugged the wrong cable into the wrong speaker at best it might not work, at worst you might end up might short-circuiting the amplifier.

Canon EP8

The EP8 was bulky and probably too expensive for regular usage, but was popular on higher end multi-way speakers such as the Turbosound TMS3 which was a 3 way active box driven by 3 channels of amplification (high, mid and low)

Jack connectors.

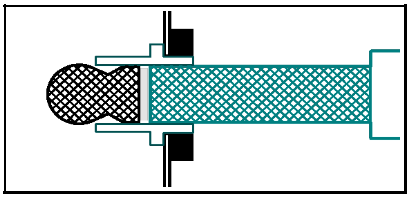
While the jack connector is still used on instrument amplifiers and cheap speakers today, I consider the Jack connector to be the worst possible choice for a speaker connector (particularly on higher power systems) for several reasons.

1) **They are potentially dangerous.**

   When the Jack is unplugged, both terminals are exposed and when used with an amplifier capable of 625Watts (into 8 ohms) there may be over 100V peak on the contacts so anyone touching the connector could get a nasty shock.

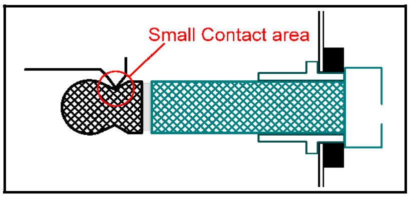
   This can also be a potential fire hazard.

2) **The amplifier can be short circuited while being plugged in**

   The body of a jack socket is often metal which will short circuit the jack as it is being plugged in.

3) **They have a small contact area**

   This means they have low current capacity and the connector cannot handle much power which will lead to the premature failure of the connector.

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Really the only thing Jack connectors have going for them is that they are (relatively) cheap, but it is pretty safe to say that anyone using them for speaker connections is making a product down to a price rather than up to a standard.

Speakon

Originally introduced by Neutrik ™, these were designed as dedicated speaker connectors from the beginning, to avoid many of the problems of other connectors.

They come in two main types: the NL8 which has 8 contacts and the NL4 which has 4 contacts although there is also a variation of the NL4 called the NL2 which has only two contacts but is otherwise compatible with the NL4.

Power Cables

OK, what can I say about Power cables?

Probably not a lot, as there will be variations from Country to country lets just say there are power cables and there are cheaper power cables – unfortunately copper costs money, so cheaper power cables are likely to have thinner wires which is far from ideal.

I live in Australia which is a 240 Volt 10 Amp per circuit country, cheaper extension leads are lucky to handle 7.5 Amps without distress and it is generally safer to purchase 'tradesman grade' extension leads these will not be cheap, typically costing $2 or $3 per meter against $1 or less for a cheaper lead.

While Tradesman leads usually come in an unfortunate choice of colours (bright colours like yellow and orange) they are usually rated at a Genuine 10 or 15 Amps, have a thicker, more rugged sheath and provided you do not treat them the way the average Tradesman does they should last you a lifetime.

Why extension leads should be uncoiled before use

This is one of those things that has accumulated its own strange mythology – you don't go far without hearing it claimed that extension leads need to be completely uncoiled before use, because Inductance will cause it to overheat and melt (a claim I have heard many times).
while it is true that you should uncoil extension leads (particularly if they are to be used to supply a heavy load) the reason is much simpler than blaming mysteries of inductance.
The problem with the inductance theory is that inductance depends on the magnetic field around a current carrying wire, now if an extension lead was a single wire I might accept this theory but an extension lead contains both an 'Active' (think of it as supplying current to the load) and a Neutral (or return – think of this as returning the current from the load) the current on these wires is both equal and flowing in opposite directions so the magnetic fields from these wires largely cancel each other out.
The reason coiled up cables overheat is actually much more mundane.

When a wire carries current it heats up, the more current, the warmer the cable gets, In fact the formula for calculating how warm a cable gets is $P=I^2R$ where P is power (heating), I is the current flowing in the cable and R is the resistance of the cable – this means if you halve the resistance of the cable you halve the heat but if you double the current in the cable you quadruple the heat.... but wait! You can make it worse!

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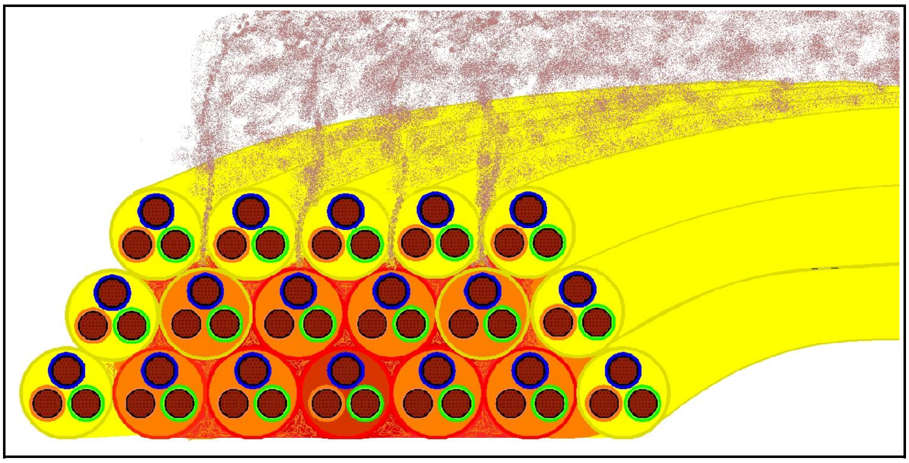


Cross section of neatly coiled cables getting slightly warm

When a cable is completely uncoiled, the heat radiates into to the air around it, cooling the cable. However if you leave the cable in a nice neat coil the outer layers of the cable insulate the inner, layers allowing heat to build up, in extreme cases the cable can heat up to the point where the insulation softens or melts this in turn can cause the cable to short circuit, drawing more current and leading to…. Well, ultimately a merry little blaze!

While we are discussing cables it is probably a good idea to cover the questions of what makes up a good cable I just touched on one factor, obviously the lower the resistance of the cable the less heating also the lower the resistance the less voltage drop across the cable. The problem is in order to get lower resistance you need more copper – half the resistance needs twice the copper, more copper costs more (and of course more copper weighs more) and the other problem is thicker copper is less flexible.

Better cables are made with more strands of thinner wire keeping them more flexible and less prone to fatigue, cheaper cables are made of fewer strands – cheaper, cheap cables use fewer strands of thin wire making them very delicate, better, cheap cables are made of fewer strands of thicker wire making them less delicate but also less flexible.

You may often see Cable described as "install cable" this is usually cable made of fewer, (and often thicker) strands on the grounds that as flexibility is not a pre-requisite for install cable this makes the cable less expensive to manufacture.

Microphones

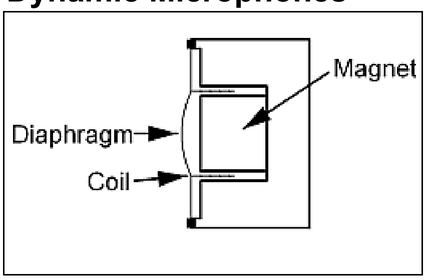
While I did start by discussing cables, microphones are where it all starts really.

Microphone types

All microphones operate on the basic principle that variations in air pressure (otherwise known as sound waves) cause a diaphragm to move. The movement the Diaphragm is then used to generate an Electric current.

Because the changes in air pressure involved are very very small microphones need to have very light weight, delicate, compliant diaphragms.

Dynamic Microphones



Dynamic microphones employ the same principles as a speaker - only in reverse.

Attached to the diaphragm is a coil of very fine wire often thinner than a human hair. this coil is suspended in a Magnetic field so that as the coil moves a current is induced

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in it. The output from this coil is then fed back to the Microphone pre-amplifier at the desk (although many Microphones feed the signal through a matching Transformer first).

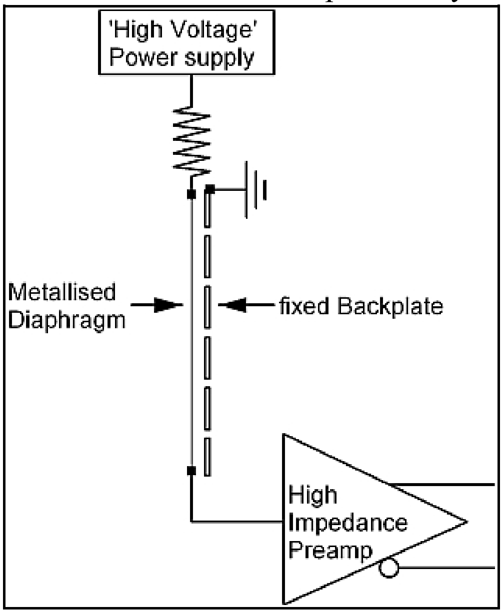
There have been a number of attempts made to make transformer-less Dynamic microphones and while some have succeeded others have had a reputation for being unreliable – one of the reasons for this is that in order to generate enough drive they have used coils with many turns, in order to keep the coil light the wire used has been very thin (thinner than normal) so it has been weaker and more prone to breaking.

Other than that issue, a well made Dynamic microphone is generally about as rugged as you can get and this has made Dynamic mics the mainstay of most (if not all) PA systems around the world.

Besides the usual suspects (moisture, corrosion, being run over by a truck etc) there is little that can be done to destroy a Dynamic microphone and offhand I cannot think of anything beyond say a very violent shock.

Condenser Microphones

OK the Condenser Microphone may be a bit tricky to explain so try and bear with me here.



A Condenser microphone consists of a tensioned metallised diaphragm in very close proximity to a fixed metal plate these form a Capacitor (otherwise known as a Condenser).

The Capacitor is charged up from a relatively high voltage (usually 48V from the phantom supply although I have heard of fancier Microphones that run off higher voltages) called the bias voltage fed via a very high resistance usually in the order of Gig-ohms.

As the air pressure causes the diaphragm to move in relation to the fixed plate this varies the capacitance between the two, as the charge remains constant this causes the voltage on the plate to vary.

Since this is an extremely high impedance system almost any load at all will swamp the signal which means it cannot be connected directly to a regular mic pre-amp so Condenser Microphones usually contain a FET (or Valve) pre-amplifier often powered by phantom power from the mixing desk (which also provides the 'bias voltage').

Weaknesses – diaphragms used to be considered delicate but most modern condenser microphones nowadays use metallised plastic diaphragms and these are generally pretty rugged.

Moisture, dirt and humidity can cause condenser microphones some difficulty as this can cause (electrical) leakage of the charge on the diaphragm which will cause crackling sounds, or even complete failure of the microphone.

Another common cause of major crackling from a condenser microphone is a faulty lead or connection – although I will cover this in more detail in 'how to corrupt your cables'

Ribbon Microphones

A Ribbon microphone is a form of dynamic microphone however unlike the dynamic microphones we discussed earlier these are extremely delicate and are almost never used in PA systems so I am tempted to end the discussion here.

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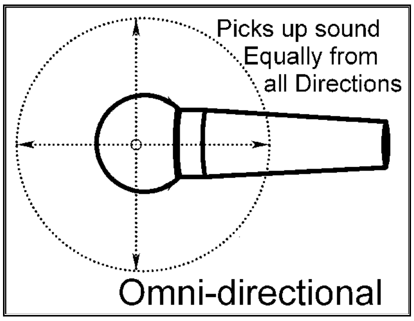
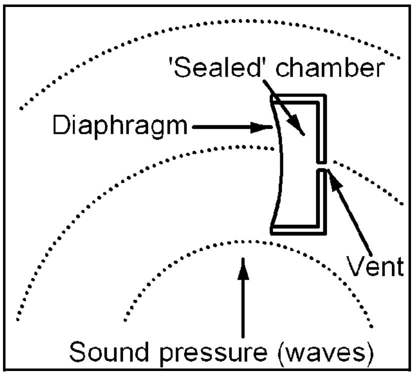
Instead of a coil attached to a diaphragm (as with the Dynamic microphone) a ribbon microphone effectively combines the two by using a very thin ribbon of (usually) foil suspended in a magnetic field. As the ribbon moves in the magnetic field this generates a current, unfortunately the voltage generated is not very high so this is fed through a transformer or pre-amplifier.

Weaknesses? Well to get the best sound the ribbon needs to be very thin and very light which in turn makes it very delicate and I have heard of ribbon microphones that were (reportedly) damaged when the box they were in was closed quickly.

Microphone patterns

Omni directional Microphones





An Omni-directional (or Omni for short) is a microphone that picks up sound equally well from all directions.

An Omni-directional microphone is essentially a diaphragm with a sealed* chamber behind it.

When the air pressure in front of the diaphragm varies with respect to the pressure in the chamber this will act on the diaphragm causing it to move.

Since this Microphone detects variations in air pressure it is classed as a 'pressure sensitive microphone'.

* In reality the chamber is not completely sealed but has a pressure equalisation vent (i.e. a small hole) designed to let air in and out slowly due to atmospheric changes without letting sound waves through.

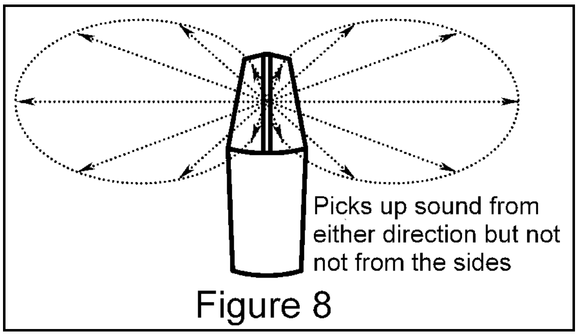


Figure 8 Microphones

Figure 8 Microphones pick up most sound from the front and rear of the diaphragm and no sound from the sides.

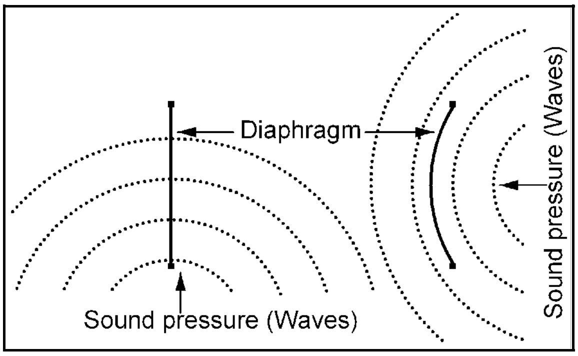


Figure 8 microphones simply consist of diaphragm open at the front and rear.

Sound waves from the front (or rear) cause a change in pressure across the diaphragm (hence the term 'pressure gradient microphone') this causes the diaphragm to move.

Sound coming from the sides of the diaphragm result on the pressure being equal on both sides of the diaphragm so it does not move.

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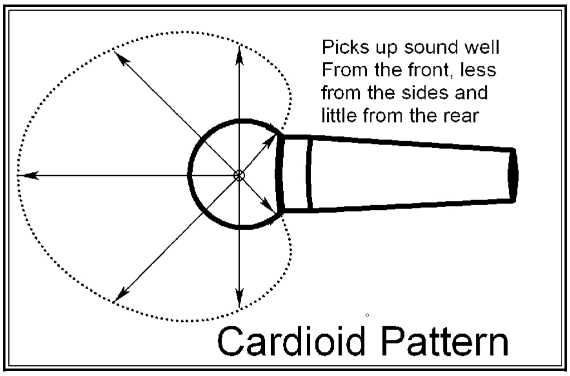
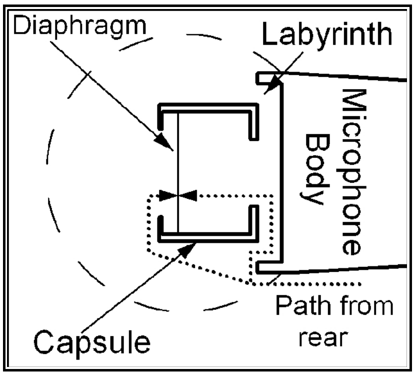


Cardioid microphones

Cardioid microphones pick up most sound from the front, less from the sides and very little from the rear*.

Conceptually (and the reason I covered it last) the Cardioid microphone can be thought of as a combination of an Omni-directional microphone and a figure 8 microphone.

A signal coming from the rear of the figure 8 capsule will be 180° out of phase so when this added to the signal from the omni-directional capsule it will cancel out any signals from the rear.



Variable pattern microphones often employ this technique – switching the omni and figure 8 capsules in, or out as required.

Fixed pattern microphones however only contain a single capsule so they operate slightly differently.

A Cardioid microphone consists of a diaphragm mounted in front of a chamber often referred to as a 'Labyrinth' the purpose of the labyrinth is simply to ensure that sounds from the rear of the microphone travel the same distance to reach the back of the diaphragm as the front with equal pressure on both sides of the diaphragm it does not move. Signals from the front of the microphone travel much further before they reach the rear of the diaphragm resulting in a pressure gradient (note the illustration here is only conceptual in nature).

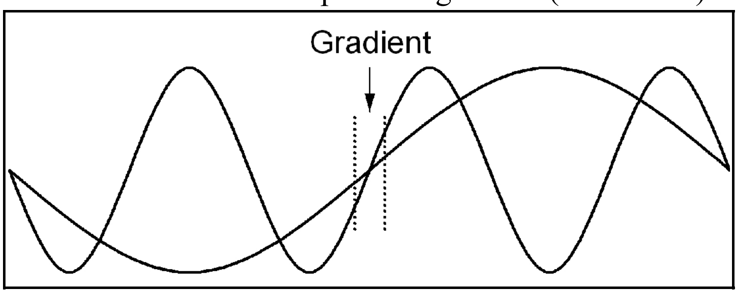
This leads to a couple of oddities of Cardioid microphones:

First if you block the vents into the labyrinth at the rear of the microphone it ceases to be as directional – most vocal microphones have a mesh ball covering the vents to discourage this sort of behaviour but you can still compromise their directional characteristics by cupping the microphone in your hand.

Second directional microphones (including Cardioid Microphones) suffer from something called 'proximity effect'. This is where a microphone exhibits an increase in low frequency response when it is picking up a sound source nearby compared to picking up a sound from further away.

While the reason for this is fairly straightforward, it does involve two factors which may take some explaining so bear with me while I run through them:

As I mentioned earlier a pressure gradient (directional) microphone relies on a difference (or gradient) in pressure across its diaphragm to work.



Lower frequency signals have a longer wavelength, this results in a smaller pressure gradient across the diaphragm than a similar level higher frequency signal.

This results in a 6dB per octave low frequency roll off, which the microphone manufacturer then compensates for.

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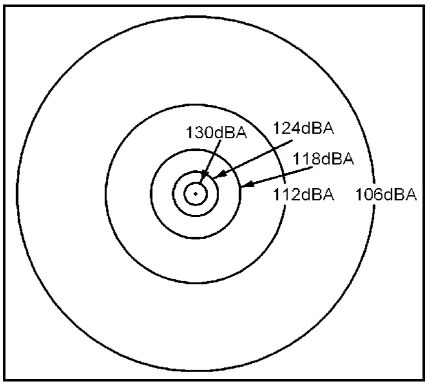
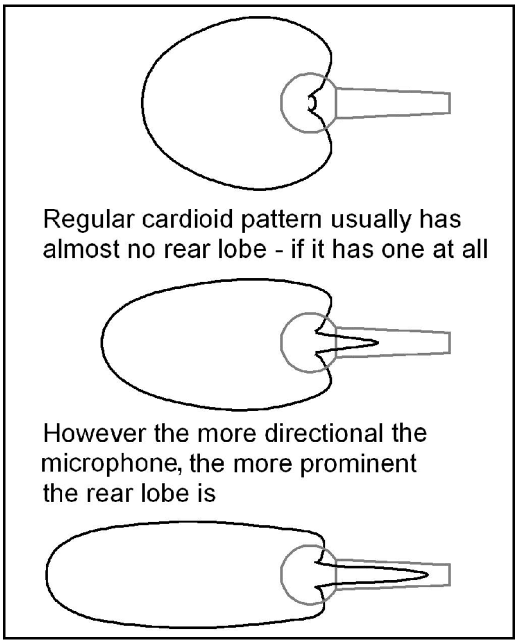


The second factor is that as a sound spreads out from its source, its energy gets spread over a larger and larger area (i.e. things sound quieter from a distance – but you know that didn't you?) this follows a rule called the inverse square rule, for the purpose of this discussion we can simplify this rule by saying that as you double your distance from the source of a sound the volume drops by 6dB.

So if your microphone is within one centimetre of the source of the sound (i.e. a vocal Microphone held up to a singers mouth) and the path to the back of the diaphragm is an additional centimetre this means that you will have (at least) a 6dB gradient across the diaphragm before you even consider the phase shift.

As a comparison a signal from 20 centimetres away will have a negligible decrease in level at the rear of the diaphragm.



In practise this means that a directional microphone will effectively have a low frequency boost when used to pick up sounds close to it in comparison with sounds from further away – this is known as the proximity effect.

Finally, before we finish with Cardioid microphones I need to mention the infamous 'rear lobe'. A normal cardioid pattern microphone will pick up nothing (well ok, not really nothing more like not as much) from directly behind it, however as microphones become more directional, they tend to have a rear lobe (where they become more sensitive to sounds directly behind them) with this lobe becoming more pronounced as the microphone becomes more directional.

DI's and other inputs

Active or Passive? ~

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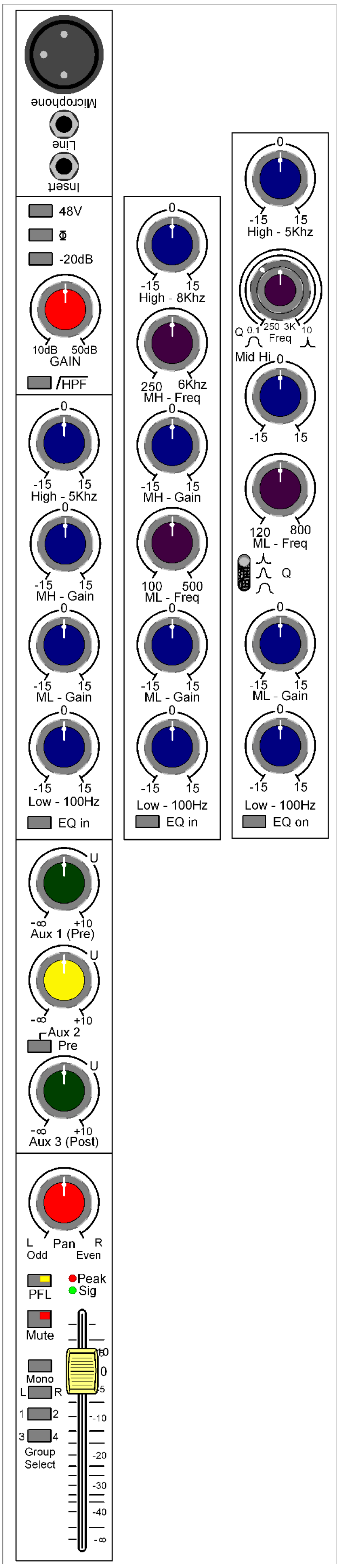


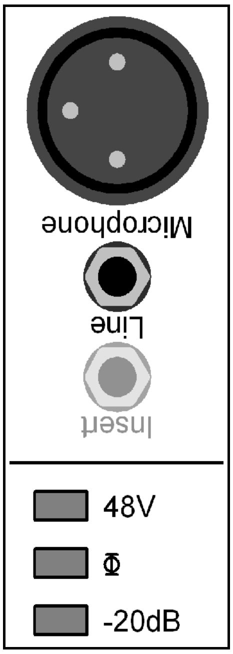
Mixer

The mixing Console is the heart of any PA system, everything goes through it and this is where the system Engineer does most of his or her work so it is worth taking a good look at a typical desk.

Note that I have tried to describe controls found on a typical console here, however your desk may vary from this, so check the instruction manual for your console.

Microphone Channel strip

Input section



This is where signals enter the console, either via the XLR (or Microphone) input or via the line input (6.5mm Jack). This is usually an either/or arrangement as they both use the same pre-amplifier with the only difference being that the line input is usually padded so as not to overload the pre-amplifier.

48V (aka Phantom) :

Selects phantom power for powering active DI boxes or condenser microphones.

Phantom power is only wired to the XLR input, as putting 48V on a Jack input is asking for trouble and can cause damage to equipment. - Note that many desks have Global phantom switches which will activate phantom power for either a group of channels at a time or all the XLR inputs – check your manual for further clarification.

On most modern desks the line input is also balanced – using a TRS jack input where Tip is in phase, Ring is out of phase (this really should be 'inverted' but we will follow common usage here) and Sleeve is ground.

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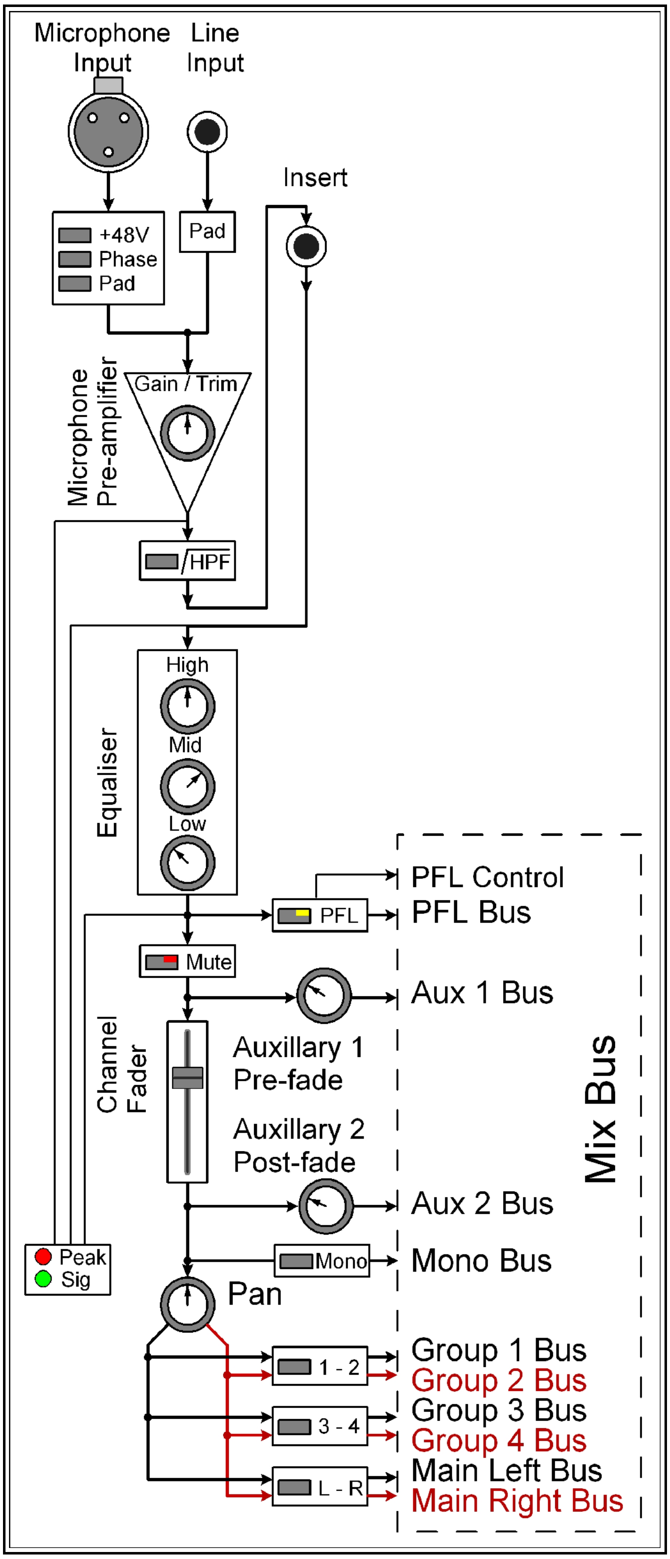


Φ (aka 'Phase' or 'Invert') :

Inverts the signal, this can be useful for dealing with 'comb filter' effects when using multiple microphones – contrary to common belief it is generally not a lot of use for dealing with feedback.

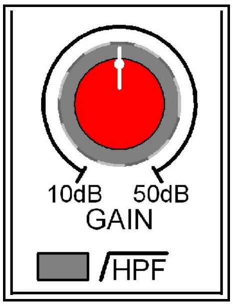
-20dB (aka 'pad' '-10dB') :

Drops the level of the signal if it is too high for the pre-amplifier to cope with.

While this example shows -20dB this may vary depending on the console.

Pre-amplifier

The Microphone Pre-amplifier is designed to cope with a wide variety of input signals and usually boasts a variable gain of anything up to 40 or 60dB.



Gain (aka 'Trim') :

This adjusts the gain of the input stage

The electronics after the Microphone pre-amplifier are function optimally higher levels (nominally 'line level') so this is why it is so critical to get the input Gain correct (See Soundcheck notes) Controls for this stage consist of a single knob marked Gain or trim.

HPF (aka '100Hz' or '80Hz'):

Following the Pre-amplifier is a High Pass filter which is usually independent of the EQ section this is useful for reducing rumbling, wind noise etc. which could otherwise be overemphasized in a system with a Subwoofer.

I generally recommend that the High-pass filter is left on for any channel that does not require low frequency response (i.e. don't use the HPF on Bass guitar, Kick drum, floor tom etc.).

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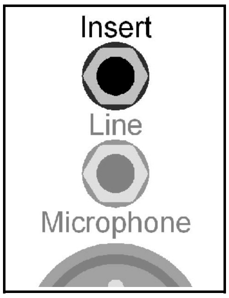
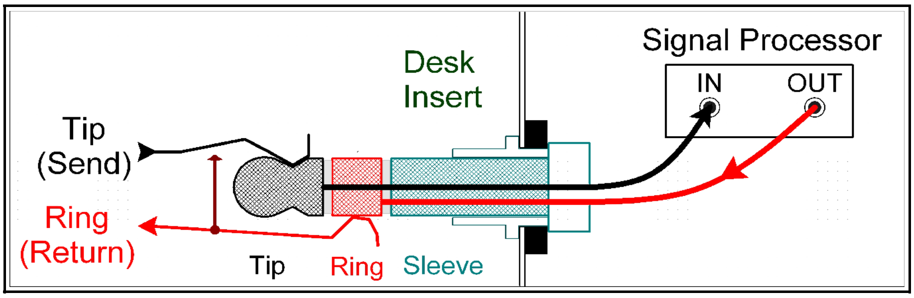
Insert

Next we come to the insert Jack – this is a socket that allows us to break the signal path through the channel strip in order to insert (as the name suggests) a signal processor such as a Compressor, Noise gate or external Equaliser.





The Insert jack typically uses a TRS (Tip Ring Sleeve) connector to carry both Send and Return signals where Tip is Send (i.e. FROM the mixer) where Tip is send and Ring is return (i.e. back TO the mixer).

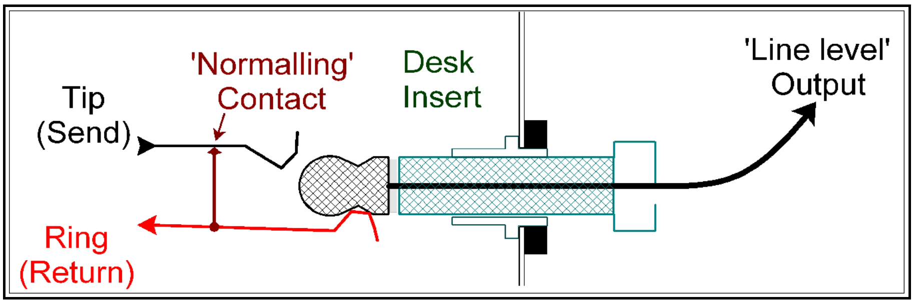
Note that while this seems to have become the standard some, some consoles (usually older ones such as for example the Soundcraft 200) may be the other way round, with Tip as Return and Ring as Send – so as always you may need to check this.

Also note the colour of Stereo to dual mono / splitter cables. while there is no hard and fast rule, typically cables that break a TRS lead into two TS plugs use the logic that Red is Ring is Return is Right.

Another tip here – if you wish to make your own TRS splitter cables, cable sold as S-VHS cable contains two individually shielded cores which can be easily and neatly split into two TS (mono) plugs.

Half normalled :

If you wanted to take a line output from a single channel on the desk (usually for multi track recording) the best option would be to use a desk with direct outs, (sorry I have not covered Direct outs in these notes but, these are line level outputs on each channel) however if your mixer does not have direct outs you can use the insert jacks instead.



The trick here is to use a normal TS lead and plug the jack half way into the socket. If you plug the jack in slowly and carefully you should feel the contacts click into the notch at the end of the jack – note that some desks will have three contacts altogether – the Sleeve, The ring and then the tip while on other desks the Sleeve is the body of the connector (So yep – you guessed it, you may have to work out which type your desk uses) – in general though you should have about 6mm (Ok, in Imperial this would be about ¼ of an inch) of the Jack plug hanging out of the socket.

This works because most desks (and yes you do have to confirm with your desk) use a normalling contact (Basically a switch that opens up when a connector is plugged in) on the Tip of the insert socket, so if your connector merely engages the ring contact the normalling contact is not opened and the desk will otherwise continue to function as normal.

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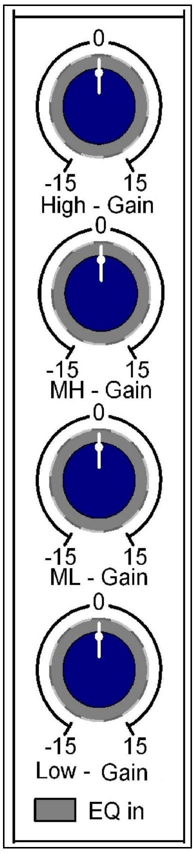
Equaliser

Shelving EQ



The shelving Equaliser is the easiest of the three (covered in these notes) to understand and use, however it is also the least flexible.

Operation is straightforward – if you want a low frequency boost you turn up Low, if you want less Highs then you turn down High and if you do not want any EQ it can be switched out using the 'EQ in' switch (in this case the button in the down position is EQ in, so up would switch the EQ out).

Limitations: Well you are stuck with whatever the designer of the desk thought you might need and in reality what you may want, will often vary from instrument to instrument.

For example Lows might be below 150Hz Mid-Lows 200 to 500 Hz, Mid-High 800 to 2.5Khz and highs 4Khz and above.

For Vocals this might very well work:

Low at <150Hz should tame plosives and rumble

Mid low at 200-500Hz is a tad broad but you should be able to control any 'muddiness' without losing too much clarity or tone and this should also allow you to warm the vocals as needed.

Mid-high 800-2.5Khz boosting here will add some edge to the vocals without getting too harsh.

Highs 4Khz and above cutting should help tame sibilance but may reduce some clarity.

If you now want to EQ an acoustic guitar however, these choices may not work quite as well

Low at <150Hz should help tame body and handling noise.

Mid low at 200-500Hz may help bring out some tone in the instrument but could also add a lot of body noise and muddy the sound.

Mid-high 800-2.5Khz cutting this should help tame pick noise (when it is strummed) but will also kill some of the tone of the instrument.

Highs 4Khz and above should help keep pick noise at bay.

OK so maybe we just need more bands on the Shelving EQ? But where would this end?

Sure a 10 band EQ would be helpful but adding 10 knobs to the board is just going to make it a bit too big – of course this could be done on a digital console but at the time of writing this most of the lower cost Digital consoles on the market are aimed at Recording and while they can be used for live work the subgroups (if they exist) and masters are typically paged with channel faders so they are not ideal (the exception to this is the Presonus Studio live, but in Australia these are currently about $5,000 whereas an Allen and Heath Zed420 (equivalent analogue) is about $1,700).

But I digress..... The point I meant to make is that a usable shelving EQ is going to take a lot of board space and often not all the bands need adjusting anyway, so what if instead of making an EQ with fixed frequency bands, you made an EQ where the operator could select the frequency band she or he wanted to adjust?

Well this thought brings us neatly to the next type of EQ commonly found on mixing consoles.

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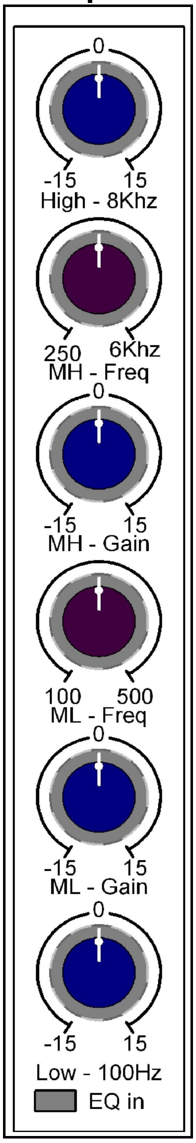
Sweep EQ



The Basic Sweep EQ section consists of two controls – Gain (aka Cut/boost) and Frequency.

Sweep EQ sections also usually also have shelving Equalisers for High and Low as well – the example on the left shows dual sweep-able mid sections with high and low shelving, as this is typical of what you may find on a mixing console.

Now I could flippantly say to use a sweep EQ simply dial up the desired frequency and apply the required cut or boost, but how to work out what Frequency you want?

As you get used to your desk, you will quickly reach the point where you can dial up pretty close to what you want, but if you are not there yet, try the following technique:

first you need to decide if you want to cut or boost

Do you want to enhance the tone of an instrument or warm up vocals?

This would probably call for some boost.

Do you want to reduce the clatter of pick noise or reduce 'muddiness' in vocals?

This would call for cut.

Cut or boost and don't be afraid to over do it – say past 3 o'clock (if boosting) or 9 o'clock (if cutting).

Sweep the frequency control around (i.e. turn the frequency knob), listen to the effect it is having on the instrument to find the frequency setting you want.

Then fine tune the Gain to get the sound you want.

This brings us to a philosophical question: Should you cut or boost?

Well Recording engineers will often say that 'you should always boost and never cut' – the reason for this approach is that they are afraid that cutting will permanently remove part of the signal and if they later decide they need that sound, then it is sometimes almost impossible to get back.

Many Live Engineers will take the opposite approach with 'You should always cut and never boost' and the reason they would give is that by boosting a frequency you are increasing the likelihood of feedback.

Me? I reckon you do what you need to do to get the job done.

Sure if you boost you increase the possibility of feedback, but then again if you cut, you may end up needing to turn the channel up to compensate for the EQ, so while feedback is always a potential risk don't let fear of feedback get in the way of good EQ.

Which doesn't really answer the question – my answer is try both (during soundcheck of course) to see what works best and after a while you will get used to which works best where and with who.

Let me give you a couple of examples:

Say you have muddy sounding vocals.

The choice here would be either cut the lower mids (around 200 – 300Hz) to remove the 'mud' or boost somewhere above 1Khz in order to add a bit of edge or clarity and pull the vocals out of the mud.

Of course if you are blessed with twin sweepable mids then you can do both, but if you only have a single sweepable mid then try one, then the other and see which sounds best.

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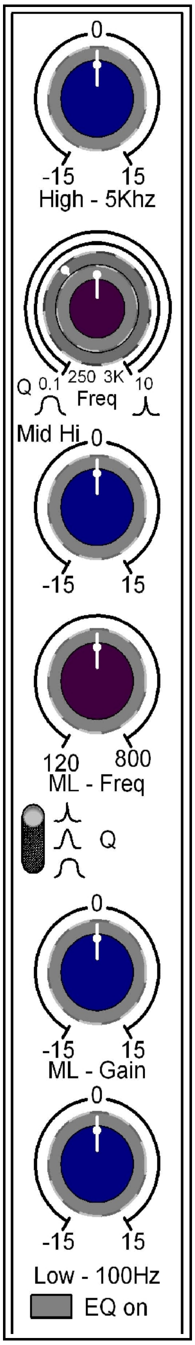
Another example:

You have an acoustic guitar which when strummed sounds very clunky with lots of pick noise.

Here the choice is, do you cut the upper mids above 1Khz to reduce the pick noise, or boost mids to bring the tone out of the instrument?



For me past experience has shown that with an acoustic guitar cutting can be problematic as often by the time you have cut enough to alleviate pick noise you have sucked most of the life from the guitar, so I would start by boosting the mids and sweep the frequency around a bit, you will often find a 'sweet spot' usually somewhere around 500Hz or so which does seem to vary slightly from guitar to guitar and also from Desk to desk.

Limitations? While the sweep EQ is slightly more complex to use, it is pretty effective, but sometimes you might need just that bit extra control...

Parametric EQ

The Parametric EQ adds another control, in addition to cut/boost and Frequency we now have a mysterious additional control called 'Q'.

Maybe to understand this, it would be easier to look at an example first so lets go back to the Acoustic Guitar mentioned at the end of the section on the sweep EQ.

You may find the frequency range of the sweep EQ too broad to easily find a sweet spot, if you sweep the frequency lower the sound becomes too muddy but if you sweep it any higher the pick noise starts coming through and the sound gets too harsh.

What you want in this situation is a way of adjusting the width of the band of frequencies controlled by the EQ – and this is what the Q control does.

Q stands for 'Quality', this is the parameter in a Filter that describes how broad the Filter is. Q factor is based on the assumption that a higher Quality filter has minimal impact on adjacent frequencies to the frequency of interest, so in other words a higher Q setting results in a narrower filter.

Rather than adding an extra knob many Q Controls I have seen on Analogue consoles come in either the form of a concentric knob (typically where the lower control is Q and the upper section of the knob is Frequency) or a switch selecting between several Q settings (both these options are shown in the picture on the left).

An EQ with a high Q can be very effective in dealing with Feedback – more so than the traditional 31 band or third Octave Graphic equaliser which has a relatively low Q when compared with what a good parametric EQ can deliver. For this reason it is not uncommon to find Parametric Equalisers on the output of Foldback consoles or even other higher end analogue mixers.

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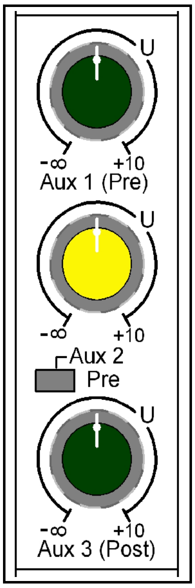
Auxiliary sends



Auxiliary sends are primarily used for foldback or effects and they come in two basic flavours:

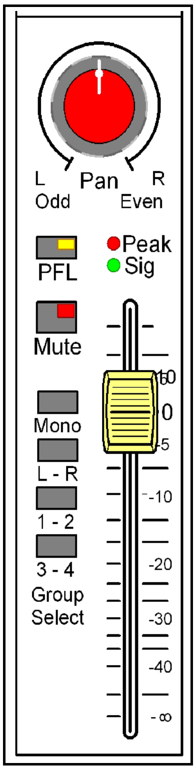
Pre-fade signals come from before the fader (hence the 'pre') and are usually used for foldback as the fader settings have no effect on the level to the auxiliary send.

Pre-fade sends come in two minor variations – pre-EQ (i.e. before the EQ) and post EQ – this can be a matter of personal preference, I like post EQ as at least you can get decent sounding foldback but the argument for pre-EQ Foldback is that changing the EQ can increase the risk of foldback – of course on the other hand having some channel EQ can also lessen any chance of feedback....

Post Fade sends are usually used for things like effects, auxiliary fed Sub woofers and recording. Post fade signals come after the fader so that as the fader is adjusted this also changes the level to the auxiliary send.

Of course the common variation of the two flavours of auxiliary send is one that is switch-able between pre or post, while most consoles are switch-able on a channel by channel basis some have a 'Global' switch controlling the auxiliary sends on all the channels at the same time..

Pan, Routing, warning indicators and Fader



Peak, Signal (aka Clip +*xx*dB, -*xx*dB) Warning indicators

The Warning indicators highlight any channels that are being overloaded or running into clip.

Peak indicators will usually light up 6dB or so before the channel goes into clip (hopefully) giving an advance warning. Signal present indicators can be very handy for locating incorrectly patched channels – saving having to PFL all the likely suspects in turn to find where something is plugged in.

One brief word on the warning indicators, high end consoles use multi-point detection for the warning indicators. This is circuitry that monitors several points on the channels strip i.e. after the Mic pre-amp, after the EQ and after any other points where the channel strip potentially provides extra gain.

PFL on the other hand usually only monitors the channel after the Equaliser section – this means that warning indicators <u>could</u> show a channel going into clip while PFLing that channel shows that the level is clear of overload.

For example (not sure why you would do this) if you had a Bass Guitar driving the Pre-amplifier into clip followed by an EQ section cutting a lot of Bass then the PFL level could show the channel is still 15dB clear of clip while the clip indicator may show the channel is in clip.

In reality this is very rarely a problem as if you have Gain set up correctly there is usually more than enough headroom to allow for normal EQ Variations – but it is something to be aware of.

Pan

This controls the stereo image of the channel and (if Left – Right has been selected) allows the image to be moved from left to right in the front of house sends. If Subgroups have been selected then Pan moves the channel between the odd and even subgroups.

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PFL (aka Solo or Cue)

Pre-Fade-Level or Pre-Fade-Listen (depending on who you ask) this switch allows the channel to be monitored In the headphones, control room out, and a level meter (usually the main level meter which should* also have an indicator to let you know it is monitoring PFL). This is vital for setting input gain and trouble shooting.

As the name suggests this lets you monitor before the fader. A common variation of this can be called 'Solo' which may be pre or post fader (Technically Solo on a recording console is post, but on live consoles is usually Pre) or SIP (Solo In Place) which is both after the Fader and after the Pan control so it lets you hear not only the level in the mix, but also the position of the stereo image.

SIP is usually only found on recording desks rather than live consoles.

*Should? - I can think of at least one popular series of console with no global PFL indicator – the Yamaha MGxx6C mixers (but there are probably other consoles out there as well).
Why a company like Yamaha (who I would have expected to know better) have done this, is unclear – especially as there are no channel PFL indicators either, which actually makes it very difficult to see when PFL has been selected on any channel.

Mute (aka 'Channel on')

Mute turns off the channel, when it is pressed it kills that channel to all auxiliary sends and to the Master outputs – on better consoles a muted channel can still be PFLed but on cheaper consoles muted channels cannot be PFLed.

A common variation to Mute is channel ON, this is like mute only in reverse, when NOT pressed it kills that channel to all auxiliary sends and to the Master outputs.

Most consoles have indicator lights to tell you when a channel is on or muted (if unclear check your manual to see which variation your desk has).

Routing or group select (L-R, 1-2, 3-4)

These switches control where the signal from the channel is sent, these usually work in conjunction with the pan control so for example if you select 1-2 and the pan control is central the channel goes to both subgroup 1 and 2 equally, if the pan control is left then the channel goes to subgroup 1 and if the pan control is right then the channel is routed to subgroup 2.

Mono (aka M, Centre or C)

Mono sends provide an extra output from the console and this can be quite handy if you are running your system in mono, want an extra output for recording, or a centre/front fill in a larger system.

There are two different types of mono outputs found on mixing desks.

Independent or assignable Mono

This is the type of Mono send shown here, as the name suggests an independent mono send can be independent of the Left and Right outputs (although these can sometimes be routed to the mono send as well) and channels and subgroups can be sent to an independent mono send without going to left and right.

Summed Mono

If the channels have no option for routing to 'mono' (as shown here) then any mono send the desk may have will be a summed mono.

This is when the left and right outputs are mixed together and sent to a connector on the desk (usually via its own level control).

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Subgroups

Many beginners are scared off by the idea of subgroups, as they think they add to the complexity of the mix and this is a shame really, because in reality Subgroups make the mix less complicated and mixing a whole lot easier.

Let me explain Subgroups like this:

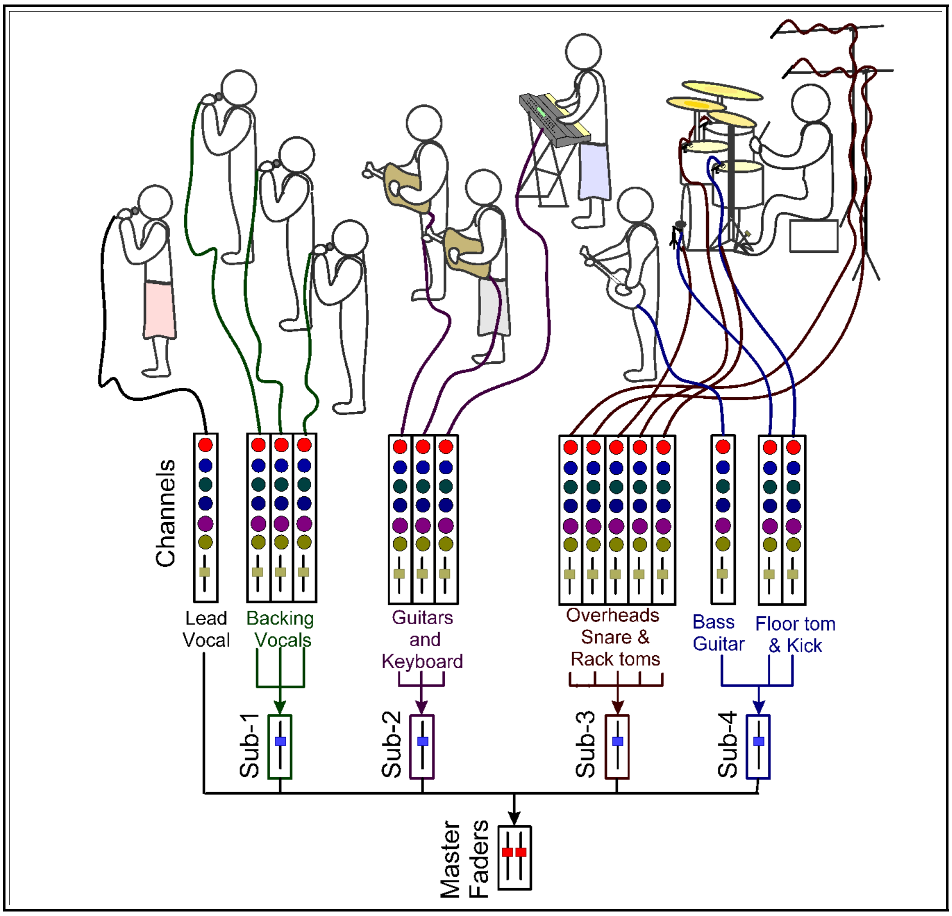
Say you get a nice balance between bass guitar and kick drum – plenty of rolling bass without obscuring the attack of the kick drum.

For mellower songs you want less bass and kick, however you want to crank them up for the more energetic songs.

Normally you would need to rebalance bass and kick each time you fade them back or crank them up. However, if you sent them both to a subgroup this would give you a single fader that lets you control both together while maintaining this balance.

When I am mixing at least half my work is done on the subgroups and once I am running a system with more than about 8 channels I really would prefer to use subgroups.

A picture of the idea behind subgroups :



This Illustration (which by the way you do not have to follow, but more on that later) shows a 4 subgroup desk with channels allocated as:

Lead Vocal	goes directly to Front of House (not via a subgroup).
Backing Vocals	go to subgroup 1.
Guitars and Keyboard	go to subgroup 2 (in this example the Keyboard is mono).
Bass instruments	Bass Guitar go to subgroup 4
Drums	Rack toms, overhead, and Snare drum go to subgroup 3.
Drums	Floor tom and kick drum go to subgroup 4

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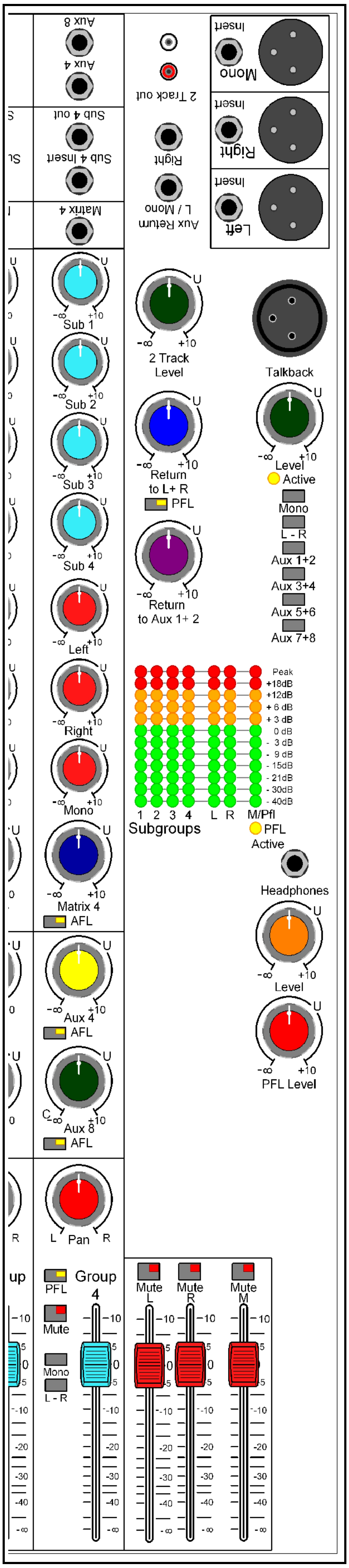
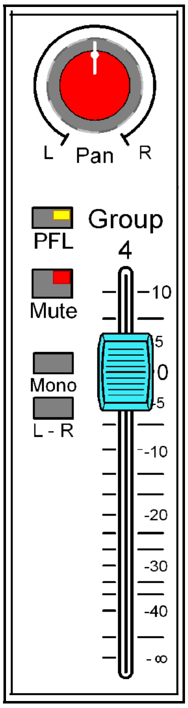
Master section and Outputs



Subgroup

We finished the input section with the group routing so we might as well follow the signal flow and look at the subgroup next.



Sub group Controls are fairly basic and simply consist of:

Fader

Controls the Subgroup level in the front of house mix.

Routing – (L-R, Mono and Pan)

This selects where the Subgroup gets sent.

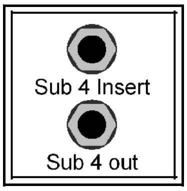
Mute

Turns the Subgroup off while leaving the fader in place.

PFL

Allows you to monitor the Subgroup (Pre-fade) could also be AFL (After Fader).

Subgroup connections:



Insert

Lets you insert a signal Processor such as a Compressor, EQ, Gate etc. into the subgroup.

Subgroup out

An output for the signal from the subgroup – this can be useful for multi-track recording.

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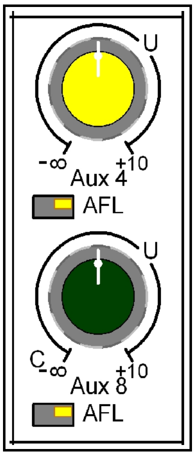
Auxiliary outputs



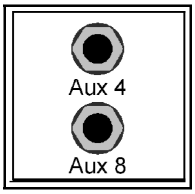
Auxiliary Controls again are pretty basic:

Aux Level

Well this ain't hard – it controls the level of the Auxiliary send

Aux AFL

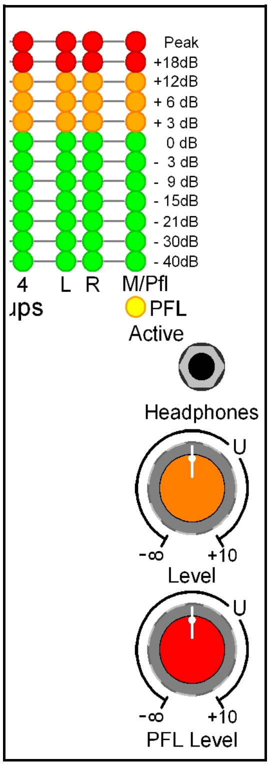
Lets you monitor the Auxiliary send – this time though it is AFL or After Fader Level so the level control also affects the level you are monitoring.



Output

There's not a lot to say – most medium to low end consoles use Jack sockets for the output (as shown here) but nowadays these are usually impedance balanced on a TRS Jack. Higher end consoles will likely have Auxiliary sends on XLR connectors which may also have insert jacks.

Monitoring



Meters

The meters show output levels, note that one or more meters will often switch between being an output meter and the PFL meter when a something is PFLed or AFLed in the example here this is the Mono meter and this is shown by the label M/Pfl.

Ideally the meters should be showing around 0dB at normal levels.

PFL Active Indicator

This warns when PFL or AFL has been selected and shows that the meter and headphones are now monitoring PFL / AFL.

PFL Level

Adjusts how loudly PFL (or AFL) levels can be heard in the headphones.

Headphones Level

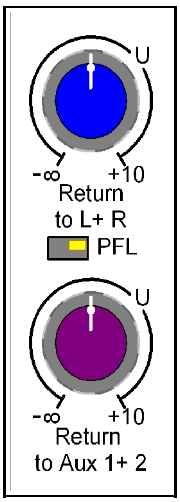
Adjusts how loud the headphones are

Headphones out

This is where you plug in the headphones .

If the desk has a 'control room output' this is usually a line level version of the headphone output.

Auxiliary Returns



Auxiliary returns are line inputs intended to take returns from effects units etc. I find Auxiliary returns are too limited to be of much use, so I generally use stereo channels (where available) instead.

PFL

Same as other PFLs, this lets you monitor the Auxiliary return

Return to L+R

Adjusts the level in the auxiliary return being sent to Left and Right

Return to Aux 1+2

Adjusts the level of the auxiliary return being sent to Auxiliary 1 and 2

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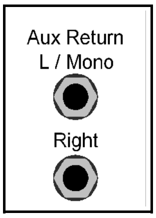
Auxiliary return inputs

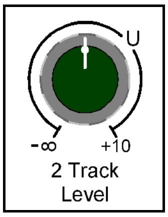


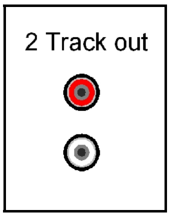
Since the auxiliary return is often stereo the input usually takes the form of two jack sockets (as shown here).

These are switched internally, so that if only one input is used, then this will be split to left and right equally.

In this example this is the left input which is indicated by the labelling of the socket as “L / Mono”.

2 Track, Record, or Alternate out



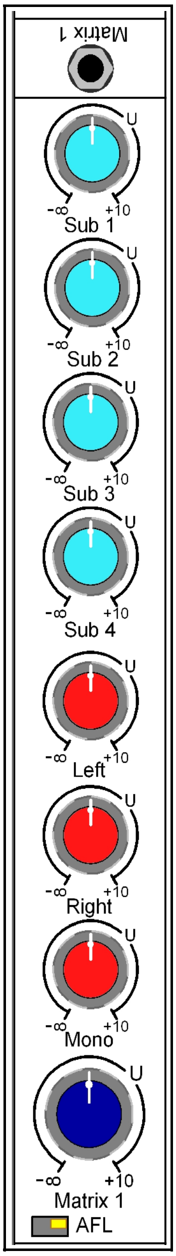


This output provides a convenient copy of the main left and right mix for recording.

While better desks provide an independent level control for this send cheaper desks may not.

There is usually no PFL or metering on this output as it is expected that any recording device will have a meter.

Matrix



A Matrix allows an alternate mix to be created from the subgroups and main outputs.

This mix can be used for recording or fill as required – for example in a recording you may want more drums etc. while you may not want these in the live mix (as the drums can often be heard clearly without reinforcement) or a centre fill may want less Guitars and Drum overheads than the main system.

While the matrix has a few knobs and may look a tad intimidating at first it is fairly simple.

Sub 1, Sub 2, Sub 3, Sub 4

These vary how much signal from each subgroup gets sent to the matrix output – Note that these are usually Post fade so that adjusting subgroup faders also varies the send to the matrix.

Left, Right, Mono

These vary how much signal from the master outputs gets sent to the matrix output – Note again that these are usually Post fade so that adjusting master faders also varies the send to the matrix.

Matrix (1, 2, 3 or 4)

This is the master level control for the matrix output so it adjusts the level out of the matrix.

AFL

Lets you monitor the matrix output – Note this is AFL.

Matrix output

In this example this is shown as a Jack – on most modern desks will be an impedance balanced TRS jack. Higher end consoles may very well have the matrix output on a balanced XLR connector.

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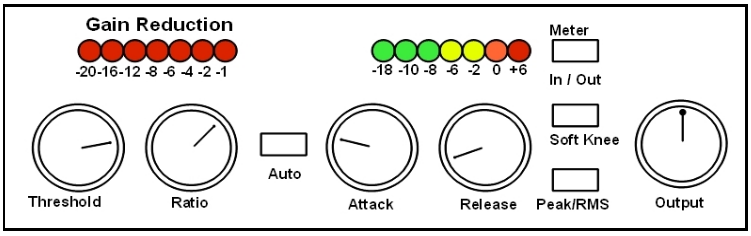
Processing and effects

Compressor



When used correctly compression can be a very useful tool for helping keep levels under control (especially vocals). Unfortunately while it is an extremely useful tool in helping maintain a balanced mix, it seems that compression is often overused and completely misunderstood.

While a compressor on each channel needing compression is the ideal setup (I would rather buy a quad 'adequate' compressor than a dual 'high end' compressor) If your budget does not stretch that far, then you can insert compressors on Subgroups. The only thing to remember if you do this is that the loudest instrument on that Subgroup is going to control the compressor.

The basic idea behind compression is pretty simple, if the level of an instrument gets too high then the compressor will automatically turn the level down. That’s it! How could anyone have trouble with that?

As is often the case, the answer is in over use, a compressor that is working hard will take the dynamics out of the instrument which in turn can suck the life out of the sound. A compressor that is working very hard may actually start to distort the Signal as well.

Ideally a compressor should be set up so that it is barely compressing at normal levels and is mainly only dealing with peaks.

Although there are often more controls, the two fundamental controls for a Compressor are Ratio and Threshold.

Controls

Threshold

Sets the level at which the compressor starts compressing (i.e. turning down the level).

Ratio

Sets the amount the gain is reduced by, when the Input signal exceeds the threshold.

Ratio is between the level of the input signal (above Threshold) and the level of the output signal above Threshold.

For example at 2:1 the output signal will increase by 1dB for every 2dB increase over the threshold in the Input Signal and at 4:1 the Output will increase by 1dB for every 4dB increase of the input signal A ratio of over 10:1 is considered to be a limiter.

Overeasy ®, soft knee, or Interactive knee

This ‘softens’ the transition into gain reduction, which helps make Compression less noticeable.

I suggest you use soft knee as a default unless you have a good reason for using hard knee (such as when dealing with percussive sounds, or using the compressor for speaker protection.

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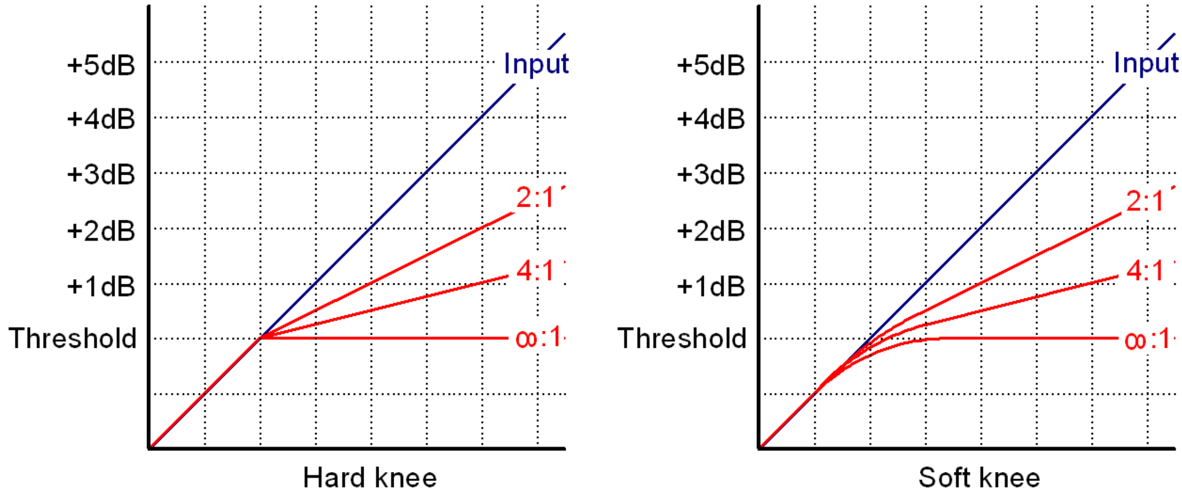

Peak/RMS

Selects if the Compressor responds to the peak signal level or the RMS (think of this as the average) Signal level. Unless the compressor is being used for a very percussive sound (such as Snare drum etc.) when you would want the compressor to respond quickly, it is safer to leave this in RMS.

Attack

Sets how long the compressor takes to respond to an increase in level. Generally keep the attack reasonably short if you want the compressor to be responsive (especially when using soft knee) If however you find the compressor is 'pumping' or causing sudden noticeable level variations then increasing attack time (i.e. slowing the compressor down) may help alleviate this.

Release

Sets how quickly the compressor recovers after the signal drops below the threshold. Setting a longer release can help keep the gain more even, making the action of the compressor less perceptible.

Output Level

Adjusts the output level out of the Compressor

Tips

For vocals start by with threshold at maximum, select a Ratio between 2:1 and 4:1, soft knee, attack and release set to auto (if available if not, set attack as fast as it will go and release to somewhere around 100mS – unless your compressor does not have soft knee, in which case you may want to slow attack down to maybe 25mS or so) and Peak/RMS to RMS.

Now adjust threshold so that at normal vocal levels, Gain reduction or compression indicators just flicker occasionally (but no more than a couple of dB).

This will set the compressor up so that normal levels get through uncompressed, while being ready to drop gain when levels increase.

Ratio of between 2 and 4:1 will normally provide sufficient gain reduction to keep the signal under control without the signal disappearing. If the signal still leaps out of the mix then increase ratio or drop the threshold, if the signal disappears in the mix then drop the ratio or increase threshold.

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Effects

Some purists may tell you that music should not have any effects, but the history of adding reverb to music goes back hundreds if not thousands of years. Cathedrals, concert halls and other venues were designed and built specifically to provide ambience and reverberation to compliment performances even in amphitheatres dating back to Greek times.

The big difference today is, that where as recently as 50 years ago a Venue was designed and built (acoustically) with one type of performance in mind (Balmain Town hall mentioned elsewhere in these notes is a good example of this, as are many Opera houses) modern electronics has meant that we can now build an acoustically dead venue and then add the acoustics that we desire electronically. This means a modern venue can be much more flexible and used for many different types of performance.

Effects can be used for a number of reasons, however with one or two exceptions the rule of thumb is usually that if the effect is clearly audible then it has been overdone.

1) As a special effect. This is usually done by, or at the request of the musician/performer and may include Pitch shifting, Delay or sampling.
2) To fill out an instrument (usually vocals). Use of a short Delay (usually around 20mS or so) can be useful in giving more body to the sound (following the rule of acoustic space this means the sound will occupy more space) while the short nature of the delay does not make it obvious.
3) To blend sounds together. Reverb or Chorusing can 'soften' a sound to help it blend in with the rest of the music. Reverb is usually the safer option as most people find it easier to understand and get working for them.

   As an example of this I often turn Reverb up on the vocals for a chorus which has the effect of blending them into the music effectively moving them back in the mix. But I will pull the reverb back for the verse in order to allow the vocals sufficient clarity to cut through.
4) To soften sounds: ok the term maybe 'soften' does not quite work here but often an instrument (particularly common with an acoustic guitar being strummed) will sound either a bit harsh or a bit bland. Reverb will help soften this and can help bring out some of the tone of the instrument.
5) Pitch correction. Although sometimes done in recording this is not practical for live performances.

Two effects I use a lot are a short Reverb with a pre-delay of 50mS and a decay time of around 500mS this reverb is used for blending vocals together and into the main mix.

I also often use a longer reverb with a pre-delay of 500mS and a decay of 750mS which provides a more spatial/airy effect and can be used for the refrain of many songs to great effect.

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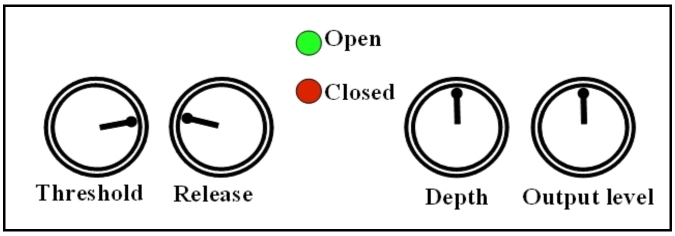
Noise Gate



The Noise gate is sort of the opposite of a Compressor – it blocks a Signal until it reaches a certain level (set by the Threshold control) at which point it lets the Signal through until the level drops below the Threshold.

Noise gates were developed to reduce Background noise behind a wanted signal – they do this by opening to let the higher (wanted) signal through and then closing when the level drops to block the unwanted background noise.

Noise gates can also be used to give percussive instruments such as Drums more attack and also help stop microphone bleed from adjacent Instruments (again usually used on drums).

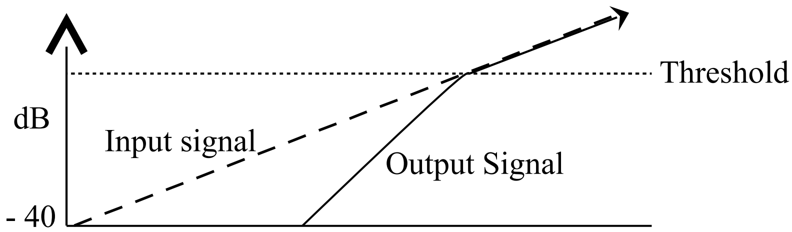
The difficulty is that it can be difficult getting the Threshold right, trying to find the balance between stopping the noise and cutting off the wanted part of the signal also the Noise gate is a fairly abrupt unsubtle effect as it is only either ON or Off which is why for live work noise gates are mainly only used on percussive instruments.

Expander

An expander can be thought of either as a gentler Noise gate or a Compressor that works in reverse… Ok that sounds like it makes no sense so let me try and explain:

With a Compressor when the signal exceeds the threshold the Gain is turned down.

With an Expander as the signal drops below the threshold the gain is turned down.



This means that the Expander acts very much like a Noise gate, however where the noise gate is only either open or closed, the expander comes on more gently.

So where a Noise gate can be very abrupt making it better suited for percussive sounds an expander is more gentle in reducing noise on less percussive instruments or even vocals.

At the most extreme an Expander will have the same controls as a Compressor and allowing for the different function of the Expander they all do pretty well the same thing. Most expanders however are much simpler and usually only have threshold and release.

Controls

Threshold

This sets the level at which the Gate opens and lets signal through.

Release

This sets the time that the Gate waits for after the signal has dropped below the threshold before it closes.

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Depth

Sets how much the signal is attenuated by when the Gate is closed – keeping the depth as low as possible makes the effect of the gate less abrupt.

Output Level

Adjusts the output level out of the Gate

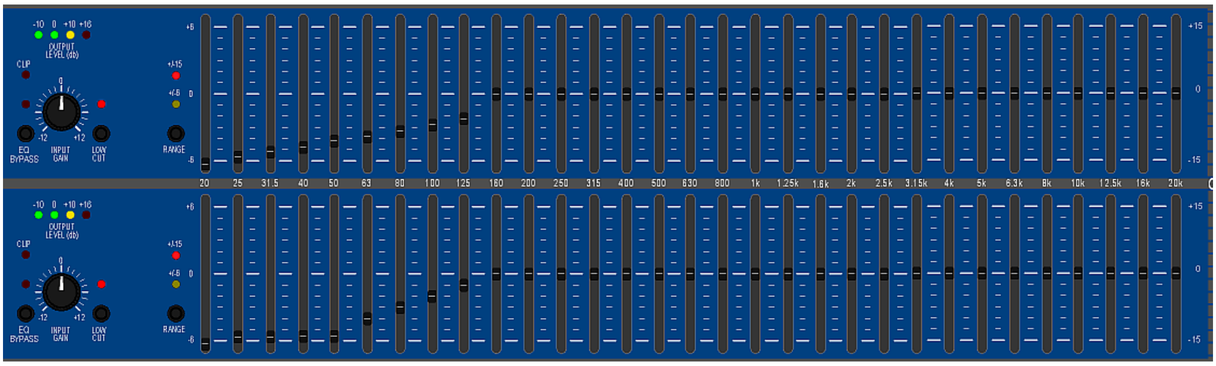
Graphic Equaliser

Graphic Equalisers are commonly found in PA system providing EQ on foldback and front of house sends. They allow the PA operator to compensate for variations in room acoustics and go some way in helping deal with feedback.

Unfortunately though the best Graphic EQs on the market can only allow you control of a 1/3rd of an octave at a time whereas a room resonance or feedback loop can result from a peak (in frequency response) of less than a 1/10th of an octave wide. This means that using a Graphic EQ to fix feedback can be a bit like cracking a walnut with a sledgehammer, you can do it, but need to be careful you do not overdo things.

Using anything less than a 1/3rd Octave EQ is even worse and generally it is advisable that for feedback suppression you only use a third octave EQ or better.

Better? - many off the shelf parametric EQ's allow much narrower filters But these usually only have a few bands and are more fiddly to use, so I would hesitate to recommend them for this purpose, especially for the less experienced operator.



Using the Graphic EQ is very straight forward there is a fader for each band of frequencies – moving the fader up boosts that band and moving the fader down attenuates that band.

Of course using an EQ to tackle feedback means that you need to be able to pick a feedback frequency while it is still building and ease the required band down now while you can train your ears* to at least isolate a feed to a few likely faders (or less) there is a useful technique that I call ‘feeling for a feed’.

To feel for a feed wiggle each fader up and down in turn, go further up each time you wiggle upwards and listen carefully for any sign of ringing as you do this (In order to hear clearly you will may need to use headphones to monitor the send you are checking).

If the band starts ringing close to 0dB (the middle of the fader) then that band is likely to be prone to feedback and may need to be pulled down significantly If however, you get no ringing until near the top of the fader (or not at all) then it is likely you will not have any major feed back problems at that frequency and you can return the fader to 0.

* there is software available off the web called the “Simple Feedback Trainer” which is a very useful tool for training your ear to pick different frequencies

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Crossovers / System 'Drive'

A Crossover is used to split the sound up into different Frequency bands so these can then be sent to different Speakers. The number of bands a crossover splits the sound into is generally termed the number of 'ways'. So that a simple crossover that splits the sound into high and low for driving a subwoofer would be termed a 2 Way crossover.

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Channel plot

When you are mixing you need to be able to find channels quickly and instinctively – there is no point having the lead guitar go into a lead break if you cannot find his/her channel to feature them until their lead is over.

The channel plot really is up to you the operator, place things where they make sense to you and you can find them for example:

I like to have vocals and lead instruments under my left hand with subgroups/masters under my right so for me a typical channel plot is as follows:

Vocals starting with any MC/Announcement microphones then Lead Vocals and Backing vocals – for the sake of simplicity the backing vocals are arranged from left to right as I face the stage.

Next I have lead instrument or instruments then rhythm or fill – I try to keep like instruments together, so for example I might have lead guitar, rhythm guitar and then keyboard.

Although the keyboard may lead from time to time, it is still easier to find (for me – set this up to suit yourself) on its own.

Alternatively if the keyboard plays a more major role I might place it before the guitars (although to be honest, I am so used to Guitars then keyboards I probably would keep it in that order).

After this I work my way down through bass Guitar, percussion and drums which I usually arrange in the order overheads, snare, rack toms, floor tom then kick drum.

So for me a typical channel plot might be:

Desk	M/C	Description		Desk	M/C	Description
1	RX 1	M/C Announcement		13	R 2	Sax 1
2	RX 2	Lead Vox		14	R 3	Sax 2
3	L 1	B/Vox 1		15		
4	L 2	B/Vox 2		16	L 9	Bass Guitar
5	L 3	B/Vox 3		17	R 4	OH L
6				18	R 5	OH R
7	L 4	Electric Guitar		19	R 6	Snare
8	L 5	Acoustic Guitar 1		20	R 7	Rack 1
9	L 6	Acoustic Guitar 2		21	R 8	Rack 2
10	L 7	Keyboards 1		22	R 9	Rack 3
11	L 8	Keyboards 2		23	R 10	Floor Tom
12	R 1	Trumpet		24	R 11	Kick Drum

In this example I am using two Multicores one on the (my) left side of stage designated 'L' and one on the right hand side of the stage imaginatively called 'R' (R usually has a bit of red heat-shrink on it just to make it clear which is which). While you might think this would increase setup and pull down time (dealing with two multicores) you usually make up for the time with shorter cable runs and less cables crossing the stage.

As I have not used all 24 channels this leaves 6 and 15 free for any last minute unexpected additions 6 of course can be either another vocal or an instrument and 15 would be an instrument.

As always this channel plot really is only a suggestion, so feel free to work out what works best for you.

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Subgroups

Subgroup allocation depends on how many subgroups your desk has but also the style of the band (and how well they play together) and what you find helpful in terms of grouping similar textures together or instruments that need to remain balanced against each other.

If we go with the channel plot we looked at earlier we have several potential groups of instruments but again these are only suggestions Subgroup allocation is up to **you** and what **you** find gives **you** best control.

Vocals

Vocals consist of two styles – lead and backing, if you have enough subgroups available you might like to keep these separate, even if you end up with only the lead vocal alone on one of the subgroups. Alternatively if I do not have enough subgroups for me, lead vocal is easy to find, so I usually send it directly to front of house.

Separation of lead and backing vocals allows you to bring the backing vocals up for the chorus or refrain and makes it easier to feature the lead vocals during the verses.

On the other hand (and this is where the musical style becomes important) if the vocals do a lot of harmonising or Acapella style singing then you may want to keep all the vocals (lead and backing) together on the same subgroup so you can maintain the blend if you decide to ride the vocals.

Guitars

Again there are two types in use here Electric Guitar most likely providing a lead role and acoustic guitars more than likely providing rhythm and or fill. Typically you may want to ride the lead guitar (or other lead instrument) to feature it as needed however its place in the mix is often relative to the rest of the instruments so while (if you have enough subgroups) you may choose to give the lead instruments their own subgroup I have no problem with allocating lead guitar to the same subgroup as the others.

Keyboards

As always it depends on how the band use the keyboard. If the keyboard gets a lot of lead type roles then you may consider giving it/them their own subgroup however as with the lead Guitar the keyboards place (in the mix) is usually relative to the other instruments so you can keep it in the guitar subgroup.

Brass section

Here the Trumpet may be the odd one out but if the Brass section play together then you need to feature them together and I would try and give the brass section (Particularly the two saxophones how will most likely play together) their own subgroup if possible.

Bass Guitar

Bass Guitar may either remain on its own (i.e. straight to front of house) or more typically be balanced against the kick drum in which case having the kick drum and bass to be on the same subgroup lets you adjust them together without losing that balance.

Drums

Drums may be run through one or more subgroup – if I have enough groups I like to run overhead and snare together, and then maybe Rack toms then group the Kick drum with the bass and floor tom.

Suggested Subgroup allocations

For me (you find what works best for you) I would probably (even I may vary what I have listed here, depending on a number of factors) allocate subgroups as follows.

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4 group desk:

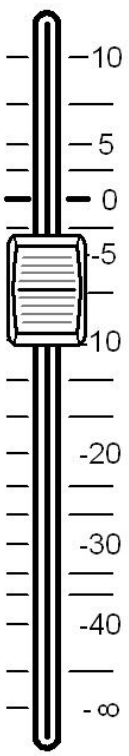
Direct to L/R : M/C microphone and maybe lead vocals
Group 1 : Vocals (this may include lead vocal), also any Vocal reverb/effects
Group 2 : Guitars, keyboards and brass
Group 3 : Snare drum and Overheads
Group 4 : Rack toms, Kick drum, bass and floor tom

8 Group desk:

Direct to L/R : M/C microphone
Group 1 : lead vocals
Group 2 : backing Vocals, also any Vocal reverb/effects
Group 3 : Lead Guitar – any lead instrument (i.e. keyboards and brass?)
Group 4 : Rhythm Guitar and keyboards?
Group 5 : Brass section
Group 6 : Drum Overheads and snare drum
Group 7 : rack toms – maybe floor tom?
Group 8 : Kick drum, bass and maybe floor tom (if not in group 7)

Where do the Faders go, in the dark, late at night?

Faders are designed to be the most readily accessible control on the mixing desk and these are ultimately where the mix happens.



Unfortunately I have heard and observed some confusion about the faders on mixing desks and where they should go, I have seen people working hard in order to keep the faders at 0 adjusting input gain and re-tweaking auxiliary sends etc. to compensate (of course if you have sound checked correctly this would not need to happen).

Faders are like the accelerator in a car, they go where they have to. Having said this however, there are areas in the faders travel that give you better control than others and you do want to try and keep the mix in these regions.

A typical mix usually happens in a range of less than 15dB, that is the lead instrument and fill instruments are going to be well within 15dB of each other with as little as 3-6dB usually being more than enough to lift an instrument up through the mix into the lead. What this means is, that you need as fine control as possible from your fader.

If you look at the picture of the fader on the left you will notice that at the bottom third of its travel (i.e. below -30dB) a small amount of movement will make a significant difference in level, so you want to keep your mix out of this region (moving a fader here to turn a channel off is fine, you just don't want to be actively mixing down here).

Above 0dB is good to keep in reserve for when you really need extra gain so this means ideally your mix should be happening between 0 and -30 on the fader.

If the gain structure on the mixing desk has been set up correctly (i.e. the channels have been properly trimmed – see the section on Soundcheck) then you should be able to achieve the levels and mix you want, with the faders between 0 and -20.

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If you find the mix is too loud with both channel and subgroup faders in this range then turn the master faders (Stereo, Left/Right etc.) down in order to achieve the mix you want at the level you desire.

If you find that the system level is too quiet with all the faders at 0 (and the soundcheck has been correctly carried out) then turn the amplifiers up, if the level is still too low then either you have a fault, or the system is inadequate for your purpose.

The master faders are not usually actively involved in the mix so there is little loss if these were set down at -30, having said this however, if your system is producing the level you desire with the master faders down in the bottom third of travel then it is very likely that the System gain is too high which will result in the system being very noisy.

Setting System (Amplifier) level

Some confusion arises as to what level to set the attenuators on your amplifiers at. Ideally the amplifiers should be set so that you can achieve the level you want when the master meters on the desk hit around 0dB the easiest way to do this is:

1) Start with the Amplifier attenuators on minimum (usually fully counter clockwise).
2) Play a CD (or Mp3, pre-recorded music, whatever - for the sake of simplicity I will call this the CD) through the desk – PFL this channel and adjust the Trim/Gain control until the meter is running around 0dB.
3) Set the channel fader for the CD to 0dB and assign directly to Front of House (L/R or Mono)
4) Set the master fader/s on the desk at 0dB – with no channels PFLed the master meters should now be reading around 0dB
5) Adjust the attenuators on the Amplifiers until you get the level you want in the venue.

   Note that it is usually an idea to set this level 10 or 15dB louder than think you will need, as you can always turn down the master faders slightly and an empty room will usually get louder than a full one so there is no harm having a bit more gain up your sleeve when/if you need it.

Soundcheck

The main purpose of a Soundcheck is to ensure that the gain structure of the Sound-system is correct, this in turn ensures that the mixing desk operates at its optimum which keeps both noise and distortion in the system to a minimum and as an added bonus makes the desk easier to operate as the channel faders will be running in their optimal range.

A proper Soundcheck is the most critical part of putting a good mix together and is every bit as important as a Musician tuning their instrument so it is unfortunate that it is often misunderstood, gets overlooked and is not properly or effectively, carried out.

An effective soundcheck is very straightforward and surprisingly does not take much time to do.

A proper soundcheck not only makes your life easier but since it ensures the system is set up correctly, it is also of great benefit to the musicians so you need to take control and insist that you get a chance to ensure the Soundcheck is carried out correctly.

Specific soundcheck procedures for a number of desks may be found at www.atps.net in the Tutorials section. If your desk is not there, feel free to contact me at tech@atps.net and I can probably put one together (note though that this would be based on information from the user manual and the desk may have an eccentricity that I am unaware of).

Why bother?

Each electronic stage that the signal passes through will add both distortion and noise to the signal. While the electronics are designed to keep this to a minimum it can only accomplish this if the signal is neither to high or to low, in addition if Gain structure is incorrect*[1] any noise or distortion created by earlier stages will only be amplified (and made worse) by subsequent stages.

There is only one stage in the sound system that is carefully designed to cope with a very wide range of signal levels - this is the Input stage or Pre-amp in the mixing desk. In order to get the optimum performance out of the rest of the system you need to make the most of the flexibility of this stage and set the gain (Input gain, Trim) so that the signal coming out of this stage is at the optimum level.

The correct way to do this is to use the VU meters on the desk – most desks*[2] have a button marked either PFL or Solo (henceforth referred to as PFL) pushing this button routes the signal from that channel directly to the headphones (and in the case of Solo any control room outputs) and one of the VU meters. The optimum level for the signal from this stage is when the VU meter is running at around 0dB for a normal signal and not going into clip for a loud signal.

I have heard it claimed that the best way of checking the level on a channel is to turn all the other channels down except for the one you are checking and then adjust the gain until the signal 'sounds' loud enough.

Not only is this completely unnecessary but it is also going to get you into trouble! while you may get away with it 8 times out of 10, at best it is subjective and there is no way of really knowing if you are running the system at its optimum or if you are pushing the desk (microphone Pre-amplifiers) in order to make up for deficiencies elsewhere in the system so nail it down and do it by the numbers.....

It is always better (and takes no longer) to use the meters provided specifically for this purpose by the manufacturer of the desk*[2].

If your channel faders all end up at some odd level after a soundcheck this may be an indication that the gain structure elsewhere in the system is not right and this can be corrected by adjusting the master faders or gain on your amplifiers.

Anyway to get on with Soundcheck….

Procedure

Initial checks

Start by ensuring that Foldback sends on all the channels are off, all the channel faders are down and no channels have PFL set – check 'PFL/AFL Active' LED.

Set the subgroup and master faders (and auxiliary masters) at 0dB (unless you know the system and are aware of some other level they should be at).

Any inserted compressors or Gates should either be pulled from the insert jacks, bypassed or disabled (i.e. by setting threshold on compressors to maximum and sensitivity on gates to minimum).

Line check

Running through the channels one at a Time:

Unmute the channel (strictly speaking you can do the gain check with the channel muted if you prefer), press PFL, ask the musician to play or the singer to sing loudly and adjust the GAIN control until the VU meter is reading around 0dB (it will vary somewhat but usually no

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greater than +4dB should work fine – depending in the desk) bear in mind that singers will often warm up, and you may need to set vocal levels a bit lower to allow for this.

Set the Foldback levels for the channel so that the musicians who wish to hear that channel can do so. Since there is no one else playing, foldback levels may not be exactly correct at this point and will most likely need further adjustment later, so get them about right but do not waste too much time on them.

Push the channel fader up and do a quick EQ, however I prefer to do the main EQ in context with the other channels so I don't spend too much time on EQ here.

Clear PFL and continue on to the next channel….

Warm up and EQ

When you have set the input gain and a rough Foldback mix for all of the channels get the musicians to play through a couple of songs and let you know how Foldback works for them.

While they are playing: assign the channels to Subgroups as required, turn the channel faders up one at a time so that you can hear each channel through Front of house and then adjust the EQ (as required).

Ideally you want to be running your faders two thirds of the way up – at the 0dB mark as gain on the lower third usually changes very abruptly and you want to keep above 0dB in reserve for 'emergency power'.

If your channel Faders all seem too low then you need to adjust the gain on your amplifiers or bring down the Master fader as this is what it is for.

Note that some folk seem reluctant to run the master fader anywhere other than 0 on the presumption that what is good for the channels is good for the master. Now while I would agree that in an ideal world the system would be set up so that this is the case (see Setting System (Amplifier) level) sometimes it is neither practical or desirable to reset system gain, in which case it is quite OK to pull the master fader down. The difference is that you will be working with the channel (and subgroup) faders to keep the mix together so these need to be at their optimum travel – the master faders however, only serve to set the overall level so it not usually a problem setting these where they need to be.

Alternatively if your channel faders are too high and your master faders are at maximum then this is most likely because the gain on the amplifier is set too low (and the only fix for this is to reset the system gain), or the system you are using is inadequate for the job in hand.

Once you have EQed all the channels (or even while you are EQing them) it is a good idea to PFL the channels again and double check the gain setting on each channel. You may find that Vocals in particular will be louder than they were during the initial check and now require some minor adjustments.

Of course any gain variation at this stage is also going to affect the Foldback levels so you will need to let the musicians know what is going on and you may need to vary Foldback settings as well.

Once Foldback, Gain and EQ have been set and the subgroups assigned, is a good time to start setting up the Front of House mix.

Soundcheck Summary:

1) Using PFL and meters adjust input Gain so the VU meter on the desk is reading around 0dB – Remember you may have allow room for Vocals to warm up.

2) In consultation with the musicians on stage adjust Foldback levels – note that for the first run through these Foldback levels will be a starting point only and will be fine tuned later so do not spend to much time on foldback at this stage.

Also you may want to quickly check EQ – but again just rough it in as you will spend more time EQing in context later.

---- Repeat steps 1 and 2 for each channel in use on the desk ----

3) Assign Subgroups and bring Subgroup and master faders up to 0dB

4) With the band playing push each fader up one at a time and listen to each channel through the Front of House system to adjust the EQ – while you are doing this also PFL and double check GAIN (note any Gain variations will also effect foldback levels which may need correction) .

5) Route channels, set up Front of house mix and carry out any fine tuning required.

While running through steps 3 to 5 keep an eye on the band to make any further adjustments to foldback that they require

*1 Correct Gain structure is where as much of the gain as possible is in the early stages of the system with (ideally) subsequent stages having either no gain or attenuating the Signal

*2 I have only stumbled across a couple of desks without metered PFL or Solo but I bet there are others out there. A Desk without proper PFL should be avoided as it is a bit like a car without a Speedometer or fuel gauge – you can make it work but it is only likely to get you into trouble.

A place for everything

In order to place each instrument (for the sake of simplicity I will include/refer to Vocals as an Instrument) in its appropriate place in the mix you need to be able to identify what role each instrument plays to get an understanding of where it should fit in the general Soundscape you are building.

Before we get too in depth here I should point out one thing, mixing is all about building a Soundscape, the Soundscape however, is determined by what the musicians want NOT YOU, your job is to be aware of what the musicians are trying to achieve and work **with** them to ensure that this happens.

While I am on this topic however, when you are working in a venue with acoustic issues (and lets face it many suffer from fairly serious issues) you are still responsible for presenting the performers in the best possible way AND this includes advising them about any issues they face in that particular venue, if this involves advising the drummer to ease up on the cymbals and high hats or the lead guitarist to tame his distortion and if they then choose to ignore your advice then so be it, you would however be failing in your duty not to inform the musicians about any potential issues they may face.

I do not want to get too hung up on terminology and you may choose to disagree with my use of terminology below – that's fine and I do not wish to get too pedantic about it, however you do need to be aware of what is happening and where everything fits

Due to the artistic nature of Music it is impossible to set hard and fast rules about composition and what Music does or does not consist of, so there will be exceptions to almost every rule however as a General rule I think of music as having one or more of (but not necessarily all of) the following layers.

Storyline

This is the part of the song that tells the ‘Story’ or core message, usually the Verse/s.

Usually Vocals – emphasis on the lead Vocalist with Backing vocals used to fill in and back up (but not obscure) the lead. Decreasing the reverb or using a longer delay on the reverb will help bring the Storyline forward in the mix.

EQ Lead Vocals may need to be EQ'ed to give them plenty of body (so don't thin them out by cutting too much lower mid) and may need just a bit of 'bite' above 1Khz to help them cut through the mix.

This is usually the most important part of the song as it tells the Story, as a result it needs to carry above the rest of the mix. Backing vocals can be used to fill out behind and strengthen the Storyline.

Refrain

This part of the song emphasises and reiterates the important parts of

the Storyline, usually the chorus.

Typically vocals - less emphasis on the lead than for the Storyline and the other instruments can be brought up more for a fuller sound.

Can sit further back in the mix than the Storyline more with the Melody but while it can merge with melody it generally should not be overpowered by the melody. Often increasing reverb and shortening the decay time for the refrain will help blend the Refrain in with the Melody.

Melody line

Provides the Tune and direction for the music, or in the case of a Solo may in fact be the tune itself.

May be vocals, although typically Lead Guitar or Keyboards but may be other instruments such as Sax, Violin etc.

Melody needs to cut through the mix and may, in some cases may bury Fill and Rhythm as Melody is second only to Storyline in terms of importance.

It is not unusual in a competent band for lead instruments to swap the melody role during a song and while the musicians will usually vary their level accordingly you need to be alert for this and if needed adjust the mix appropriately.

Also instruments may 'converse' with, or play off each other during a song and you need to give them both enough room to lead as required.

An example of this: in a Bush band I used to mix for regularly the Violin and the Banjo would often play off against each other and swap lead roles.

Note there is usually only room for one Melody line at a time, however the melody line may consist of multiple instruments, either playing tightly together (such as in orchestral music) or where instruments are playing off against each other in this case the other musicians (Frills, Rhythm etc) will need to give the lead instruments room to play and/or you may need to move them back in the mix.

Frills

may not necessarily follow or add to the melody but usually serve to provide ambience to the music.

Typically cymbals, chimes and simple keyboard sounds although may also include guitar, woodwind and brass.

These should not happen too often in a song and will usually be fairly brief so they can be featured more than Rhythm or Fill.

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Rhythm

Works with the Beat to help provide Tempo for the music, will often serve as a tie in between the Drums and other instruments – I was debating wether or not to combine fill and Rhythm as Rhythm will often also provide Fill behind the lead instrument. Fill however does not necessarily provide steady rhythm.

Typically Bass, Acoustic or second Guitar may also be Keyboard.

Usually sits just behind melody without obscuring Vocals or melody, needs to be loud enough to provide a sense of rhythm however may be allowed to blend into the fill.

Fill

Fills in behind Melody to help provide continuity to the music, often more of a sweeping/ less defined sound than Rhythm.

Typically sweeping Organ/Keyboard type sounds, may also include second guitar etc.

May be largely obscured by Rhythm and Melody sections as it provides more of a background – there can however be a couple of layers of fill where, for example brass or woodwind can take more of a leading role and fit between rhythm and Melody. It depends on the music as to wether fill leads rhythm or rhythm leads fill…

Beat

Provides the basic Tempo for the music

Typically Kick Drum, Toms and other forms of Punctuation such as Snare drum, high hats, etc.

Beat is usually fairly brief in Nature so it can usually be allowed to cut through without obscuring the rest of the mix.

Foundation

Provides Bottom end, warmth and 'Drive' for the music – care needs to be taken so that the Foundation does not obscure the beat, but rather fills in behind the beat and adds musical direction to the bass section.

Typically Bass Guitar, some organ sounds etc. Often also includes Kick drum

Due to the Low frequency nature of the Foundation there is not usually too much danger of it obscuring the rest of the mix but it does need to allow the Beat enough room to get through and also may need to be balanced against the nature of the song – i.e. a lot of foundation may be inappropriate for a gentle melodic song.

A Technical Interlude

85+85=88?

Before we get on to actually putting a mix together I need to talk about a few things regarding hearing.

The human ear is capable of coping with an extremely wide range of SPL (Sound Pressure Level – a measure of acoustic power), from being capable of hearing a pin drop several meters away, through to a jack hammer generating over 10,000,000 times the SPL of the pin.

The reason we can cope with such a wide range of sound is that our hearing responds in a Logarithmic manner. What this means is that doubling the acoustic power does not make a sound twice as loud (in fact it is considered that twice the acoustic power is the smallest

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increase in sound level that most people can detect) in order to make something sound twice as loud we actually need to increase the acoustic power by ten times.

This is why sound levels are measured in a Logarithmic scale called Decibels (a tenth of a Bel named after Alexander Graham Bell an Engineer who did a lot of work investigating the nature of hearing and who is credited with the Invention of the Telephone and the Record player).

Stating a level in dB on its own is actually fairly meaningless as all dB indicates is that you are talking about a logarithmically scaled measurement of power.

For example the VU meters on your mixing desk give an indication of levels within the desk – and while these give you an idea of changes in level they are usually calibrated in dBu where 0dBu is 0.775V (or the voltage required to produce 0.001 Watt in a 600 ohm load - This explanation is for the Geeks among you and is only important here in that it provides a context for the readings provided by the meters on a mixing desk) so while the VU meters can give an idea of sound levels in the PA system they are not a direct measurement of SPL.

Getting back to mixing, what this means is that turning up two instruments to the same level means that the resulting level sounds barely louder than with one instrument alone, also and most importantly, in order for one instrument to lead the mix it usually only needs to be a few dB louder than the rest of the mix.

In practise this means that turning an instrument up enough so that it cuts through the mix does not need to make the overall level significantly louder.

Frequency response

When putting the mix together it is also important to have some understanding of how different frequencies are heard and understood by the human ear.

An octave in musical terms is 8 notes wide (not including sharps, flats etc) Physically however an octave is double the frequency, So that middle A is 440 Hz while the next A up is 880Hz and the next A down is 220Hz.

Partly because of this and partly because of the fact that higher frequency signals in nature get absorbed or muffled readily, most of the sound we are used to hearing is below 2000-3000 Hz. If we hear a lot of power above 2000Hz or so, we find it loud, uncomfortable and very fatiguing.

In practical terms, for most people a sound with more top end sounds louder than a lower frequency sound of the same level, so while boosting top end can provide definition to a sound and help it cut through the mix, overdoing it will make it too loud and cause discomfort for the audience.

Another region to watch out for is the octave between 200 and 400 Hz frequencies in this range can either add warmth to a sound or if excessively used make the sound muddy and boomy.

The two octaves between 40 and 160 Hz can add punch to a Kick drum, power to a Bass guitar or unwanted plosives, popping and wind rumble from vocal microphones.

Note that a limitation to what you can do with Kick drum and bass will be your speakers – just because on paper a speaker might go down to say 50Hz does not mean that speaker can deliver high levels of sound below around 120Hz or so. There are many speakers on the market that claim to go low and at 1 or 2 watts they mostly do, however if you tried to run a kick drum through them at any appreciable level they would not cooperate*[2].

And everything in its place

Now we have an idea of what the mix consists of we need to give some thought as to how to put it all together

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First run through the sound-check procedure, as per the notes on doing a soundcheck.

I really cannot emphasise enough how important conducting a proper soundcheck is. If the soundcheck has been properly carried out, then it makes the rest of putting the mix together much more easy and means that the PA system will be working with you instead of against you.

Establishing a Baseline

Immediately following the Soundcheck is a good time to get the Base-line mix together and there are three approaches to setting up the initial mix the all of which have their merits and weaknesses.

The mid-way mix

The first involves starting all faders around midway and adjusting from there. This approach works well when you are not familiar with the band or their music and it lets you get a rough feel for what is happening, before establishing the baseline mix. However this approach can be a lot more work than either of the layered approaches in getting the Balance right.

Building from the bottom up

The second is to build the mix from the bottom up. This is the first 'layered' approach and involves starting with the Foundation and Beat then building the other layers on top with the other instruments.

While this works well and tends to be the most intuitive approach to setting up the mix, it can be harder to judge how loud the final mix will end up, making it very easy to end up a lot louder than you want. This is why for a live mix and particularly for a church mix I often use the third approach.

Building from the top down

The third approach is to build the mix up from the top down. This is the other 'layered' approach but involves starting with the Vocals and melody, deciding where they need to be and fitting the other parts in underneath. This method can be a bit more of a juggling act initially than building from the bottom up however I find that it does make it easier to get a feel for how loud the mix will be and makes it easier to keep the level under control.

Example

For the sake of simplicity the following describes building the mix from the top down.

The Storyline is usually the most important part of the mix, so start by bringing this up to the Level you expect to run the system at. Allow lead vocals to lead and backing vocals to build in behind, reinforcing, but not obscuring the lead vocals. Listen for anything that adds a more discordant note and ease it further back in the mix.

Next bring up the melody so it sits just behind the lead vocals – usually around or just behind the backing vocals is about right, Melody may obscure the backing vocals from time to time (although this can depend on the music – in some music the Backing vocals still take precedence over the melody) – so long as the Storyline remains distinct.

Bring rhythm up next, as this provides a lot of the drive for the song and needs to be up behind the melody, although it should generally avoid obscuring the Backing vocals.

Fill as the name suggests fills in the gaps behind Rhythm and will sit behind and may be obscured by Rhythm.

Beat is usually allowed to cut through Rhythm and may reach similar levels to the melody however Beat should be fairly brief and hopefully in time with and accentuating Rhythm so it should not really obscure either rhythm or melody.

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I have included Kick drum above in both Beat and Foundation as it may fit in either or both. Kick drum will provide the fundamental beat that ultimately drives the music, however it also usually provides a lot of the bottom end for the foundation. The Kick drum could really be thought of as Beat for the Foundation – Generally it should cut through the Foundation with the foundation acting as fill for the bottom end.

While setting up the baseline keep an ear out for anything that stands out from the mix and adjust as required.

Once the Baseline is established you can listen to the instruments in context (i.e. as part of the mix). You will often find that EQ needs additional adjustment at this stage. If you are not really confident with the EQ it may be better to leave the EQ work until you have established the Baseline mix and can EQ in context (i.e. through the PA with the other instruments).

Run through the channels one at a time to check that you can hear each instrument in the mix and that it is in the right place acoustically.

Sometimes it can be a bit tricky to locate an instrument in the mix – this can be especially true for some of the more subtle effects or where two instruments are blending together.

A very useful trick for locating an instrument in the mix is to PFL it and then place the headphones over one ear while listening to the FOH mix with the other. This will enable you to 'Synchronise' your hearing to that instrument.

Maintaining the mix.

From here there are two different Techniques for maintaining the mix.

The Static mix

The First is the static mix. This is where you balance everything as best you can (i.e. establish the Baseline) and then excepting for minor corrections, you basically leave the mix alone. This method presumes that the musicians will mix themselves.

The advantage of this methodology is that you do not have to know or be particularly aware of what the music is doing, and this will often be the most viable approach for a new or unknown band.

The disadvantages are that this method assumes that Instruments are not going to swap roles significantly during the set and that the musicians can hear what is going on in the venue and will adjust themselves accordingly (of course if this was true there would be no need for a sound engineer in the first place).

Featuring

The Second method (the one I recommend) is what I term 'Featuring'.

Before I cover Featuring in too much Detail I need to point out that a good band will to a large extent look after Featuring themselves meaning that there may appear to be very little difference between this approach and the static mix.

The difference comes more from careful listening to what is happening, making minor corrections as required and understanding what they musicians are trying to achieve.

Featuring starts by establishing a Baseline mix (as above) however adjustments are then made throughout the music in order to feature instruments as they take a lead role.

Probably the biggest advantage of Featuring is that if done with care, you can keep the level lower than it would otherwise be without the music losing any of its character and if anything enhancing the music.

Other advantages are that generally the mix sounds a lot better, it allows roles to vary and lets the music come through while also keeping the storyline intact.

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Disadvantages are that you need to be on the ball and have a good idea which way the music is going to be ready for changes as they occur, you will also need to ride instruments as you feature them to ensure that they do not get too loud which can happen if the musician plays louder to lead a song as well.

A good rule of thumb for mixing in general (and it holds true for featuring in particular) is be cautious when acting but quick to react. This means you ease an instrument up as you feature it, however keep your finger on the fader ready to pull it down if it gets too loud or you inadvertently overdo it.

You will generally find that if you are a bit slow in featuring an instrument little or no harm will be done as most folk will not notice too much. If on the other hand you push an instrument up too far it will jump out of the mix and become very obvious.

If you have a good Baseline established you should find that you do not need to push the fader up too far in order to feature an instrument and generally somewhere around 6dB will be more than adequate to feature an instrument.

Although instruments may be turned down during the mix it is usually best to avoid 'Ducking' or dropping them below the established baseline unless there is a good reason for doing so, such as an instrument adding a discordant note, too many instruments playing or when you are working in a highly reverberant Venue.

Example

In order to give an idea of how 'Featuring' works let's run through a fairly typical example:

Start the song with the Drum overheads up for the lead in to a tune (in other words start off featuring the Drum overheads – since the lead in is the drummer clicking his/her sticks together this is significantly quieter than normal overhead level and is one of the few occasions where featuring may involve going well above the base line) – as soon as the Lead in is over, pull the overheads back to baseline and feature the Lead Guitar for the Introduction.

With the introduction over, return the Lead Guitar to the baseline and feature the Vocals for the first verse. As the verse ends you may ease the vocals back and/or increase the Reverb / Delay on the Vocals and again feature the Guitar etc.

Although this is a Generic description that will not hold exactly true for every Song it does serve to illustrate the basic idea.

In the example above the drum overhead microphones pick up a lot of wideband noise from the Cymbals, High hats etc. and if we leave them up throughout the mix (as we might with a Static mix) we risk letting this noise obscure the rest of the music. If on the other hand we leave the overheads at a level where they do not obscure the mix then we will almost certainly lose the lead in to the song. As it may turn out in the mix above the Guitar will most likely not be paying until the lead over is finished so you can start the song with the guitar up.

likewise with the trade off between lead guitar and vocals, the lead vocals are the storyline in most songs. During the chorus however the Storyline is dropped and instead the vocals move to refrain which blends into/with the Melody line – this means that the Vocals move back in the mix for the Chorus and blend with the Lead Guitar. Increasing a (shorter) reverb on the vocals will also help soften them and blend them into the melody line.

***1** Melody is usually called 'Lead' however for the sake of this explanation I wanted to use a name that gave more of an idea of the role of the instrument in the Soundscape rather than its place in the scheme of things.

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***2** This part was threatening to get a bit Technical, so I figured it fitted down here better. Basically the lower the frequency a speaker recreates the more air it has to move per watt.

So while many speakers on the market advertise response down to 50Hz or less (and indeed at lower power levels they are able to manage) if you were to push them to full power at low frequencies you would find that they were unable to deliver much bottom end, will start to distort and may even be damaged.

This is why I have reached the conclusion that for general purpose work I am much better off with more compact speakers using a 12" or even 10" driver and when/if I need much performance below 150Hz or so, I add a subwoofer to the system.

The amount of air a speaker moves is a function of two things: the surface area of the speaker cone (a parameter known as SD) and how far the speaker cone can move (a parameter known as Xmax). Drivers designed for sub-woofers typically are either very large and/or (usually and) have a high Xmax with some drivers able to move 30mm or more. Unfortunately there are trade-offs involved here as drivers designed with low frequency performance in mind usually cannot deliver accurate higher frequency performance.

Other Notes

Common dB measurements

dBm	dB with respect to 1mW – that is 0dBm = 1mW
dBu	dB with respect to the voltage needed to generate 1mW in a 600 ohm load 0dB = 0.775 Volts RMS
dBA	dB with respect to 20 micro Pascals with a filter applied to approximate the Frequency response of human hearing.
dBC	dB with respect to 20 micro Pascals
dBA(leq)8h	The equivalent noise exposure (in dBA) experienced over the last 8 hours

To give you an Idea of how dB works – an increase of 10dB is considered to sound twice as loud, however it is in fact an increase in power of 10 times. An increase of 3dB is commonly thought to be the smallest increase in volume that is audible but is twice the power. 130dBA is often called the threshold of pain but really sustained sound levels of over100dBA can be pretty uncomfortable.

Noise exposure chart

Australian/New Zealand standard AS/NZS 1269 allows for noise exposure of of the following:

85 dB(A)	for up to 8 Hours
88 dB(A)	for up to 4 Hours
91 dB(A)	for up to 2 Hours
94 dB(A)	for up to 1 Hour
97 dB(A)	for up to 30 minutes
100 dB(A)	for up to 15 minutes
103 dB(A)	for up to 7 minutes
106 dB(A)	for up to 3 minutes
109 dB(A)	for up to 2 minutes

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I suggest for Church you maintain a level of no greater than 90 dB(A) – slow responding, although I would not get too carried away keeping the level down if an occasional peak in the low to mid 90's occurs. For youth oriented type services/shows I would allow the level to get up to the mid to high 90’s with the odd peak as high as 100dB(A) but there should not be any reason to exceed 100dB(A) for any long term.

Of course all these levels are going to be from the mix position and you need to bear in mind that Sound follows what is termed the inverse square rule which means that every time you double your distance from the sound source the level drops 6dB.

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