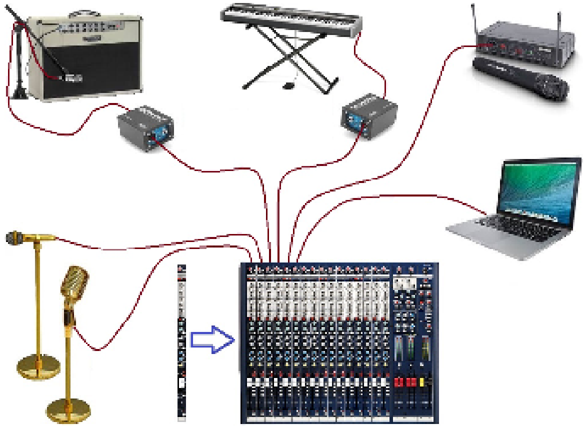
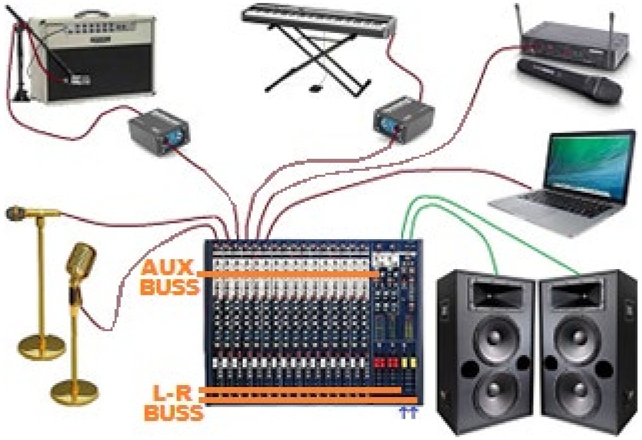

Audio Mixing Console
✝ Church Audio ✝

Audio Mixer (Mixing Console)



Channels

You should look at a mixer like a series of vertical strips, all placed side by side. We call these strips "Channels". You need a mixer that has enough channels so that all the instruments and microphones you will be using can be connected to a channel of their own. This gives you independent control of all sounds and volumes. A few spare channels are recommended for those little surprises that always come.
The channels that receive sounds into the mixer are called **Input Channels**.
The channels that send the sound out from the mixer to other devices are called **Output Channels**.



Buss

Imagine a long copper wire laid horizontally across all the channels, and then each channel connected to that wire. This is what it's like inside a mixer. Technically this wire is called a **Buss** (Buss Bar).
It allows us to send a signal from any combination of channels to a common destination.

Buss wires may be called Auxiliary Sends (***Aux*** or ***Send***). These busses are intended to send signals to auxiliary things like the Floor Monitors or Effect Units.

There is always a two-wire buss inside, at the bottom of the mixer, which is connected to the stereo output faders (***Left-Right***). The Pan knob on each channel is connect to this pair, allowing you to choose how much to send to the Left Loudspeaker, and how much to send to the Right Loudspeaker.

Digital mixers are great in the Buss department. They offer you plenty, and let you program which channels can access each one, and where they are to go to.

Connecting to an Audio Mixing Console

To keep hums and background noises out of our Sound System it is best, wherever possible, to only connect to a mixer using '**XLR**' **(3 pin)** microphone cables. Long '**Phone**' **or** '**RCA**' **(2-pin)** cables become radio antennas, picking up radio frequency hiss, cell-phone static, refrigerator motors clicking on and off, and even electrical hum if one of these audio cables runs close to a power cable.
When a 2-pin cable starts picking up noise, you need to connect it first to a Direct Box (**D.I. Box**) which will isolate the noise, and convert it to XLR (3 pin).
Very short lengths of Phone or RCA cables used on Effect Units, CD Players, Laptops or Radio Microphone Receivers are normally ok, but be vigilant.

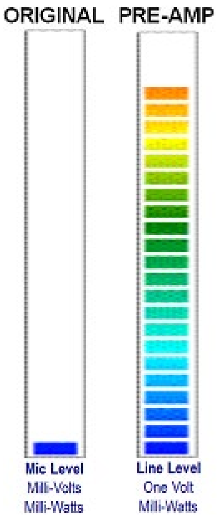
If you still hear a hum or buzz or hiss coming in to your sound system you need to systematically unplug one cable at a time to identify where it is coming from.
It may be:

i) an earth-loop caused by the Sound System being plugged in to different power lines around the building.
ii) some cable earthing (grounding) problem. Perhaps the earth wire has broken off inside a connector.
ii) a damaged (but not broken) wire inside a cable.
iii) faulty equipment which will need repairing.

When in doubt, substitute the cable to check if the disturbance goes away.

All electronic components generate a slight noise, and when you connect the many elements in a sound system together you sum all these little noises.
Consequently, it is impossible to create absolute silence, however when it's quiet a sound system should have minimal background noise (barely audible).
To test this (when it's really quiet) raise all the mixer channel faders to two thirds, then slowly raise the output Left/Right faders. Worrying?

INPUT CHANNELS

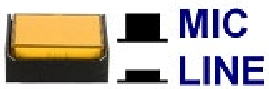


The Power Block

Mixers operate internally at an electrical level of around 1 volt. This is called **Line Level**. There is a section on each Input Channel designed to adjust the incoming signal so that it becomes Line Level.
The **Gain** (a pre-amplifier) can usually boost up to 60dB. 60dB means a "thousand times" amplification, so a milli-volt (thousandth of a volt) will become a volt. This is for microphones which only generate milli-volts.
If you connect, for example, an electronic keyboard, then the signal will already be Line Level (*because the keyboard is powered*) so you won't need to turn the Gain up very high at all.

Any time the volume from a device goes far above line level, the **Peak** light will flash to warn you. A flicker every now and then isn't the end of the world, but if a Peak light remains red for more than a moment then the signal will distort when it enters the Pre-amplifier (lowering the Channel Fader won't fix it, use the Gain).

If the Peak light stays solid red, even with the Gain turned completely down (not amplifying at all), you can press the **Pad** (Attenuation) button which will lower the incoming signal 20dB. Then you can use a little Gain to suit. Otherwise you need to lower the volume that the external device is sending you.



Selecting Line or Mic

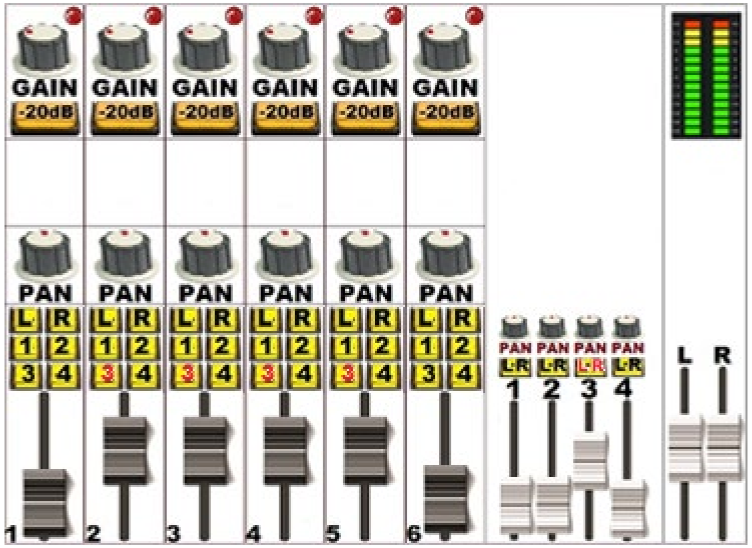
Mixers usually have a choice of Line in and Mic (Microphone) in.
On some mixers the Gain pre-amplifier is only connected to the ***Mic in***. This means that if you connected a device to the ***Line in*** you won't get any response from the Gain knob.

If you connected something to both inputs at the same time, you would theoretically hear both sounds on the channel. To avoid possible confusion, many mixers have an internal switch on the Line connector. If you plug something in to the Line, the audio coming in to the Mic connector will become disabled. Try it on your mixer.
If you have this switch-off feature it is handy if you are getting short of channels. For example during an interval you can pop the background music player in to the *Line In* of a singer microphone channel (that isn't being used right now).

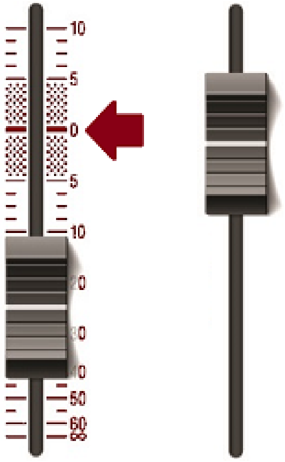
When we are using multiple microphones at close quarters, we sometimes get a weird hollow sound. This is caused by two microphones hearing the same sound, but because they are at different distances from the source there will be a delay in one. This delayed version of the signal clashes with the other (the two signals are *out-of-phase* with each other). As they mix together some frequencies will cancel each other out, and some frequencies will boost each other.
In the Power section of each Channel you may find a **Phase Inversion** button. You won't normally "hear" any difference if you press it because an audio wave is going too fast for that, but what happens is the soundwave gets inverted, so that the upward waves are now going down, and vice versa. When we hear a hollow phasing sound we press the phase-invert button on one (either) of the channels, so the two signals will be more *in-phase* with each other.
A classic application is with the drums, which have many microphones at close quarters. When in doubt, press *phase invert* and check if it sounds better.
If you are the kind that puts one microphone above, and another below, a snare drum then your two channels are 180 degrees out-of-phase. You really need to invert the phase of one of the two.

Input Channels sometimes have a **Phantom Power** button. This is used to power Condenser Microphones and expensive D.I. Boxes. A Phantom Power button may only occur every 8 channels or so, meaning that pressing one button will add Phantom power to all channels in it's group. Sometimes there is only one Phantom Power button at the rear of the Mixer, which sends power to everything. Although it doesn't bother non-phantom power microphones like Dynamic ones, try not to leave Phantom Power on when it is not needed, for electronic reasons. If you send Phantom Power to a Ribbon microphone you will damage it. Always lower the faders before pressing a Phantom Power button as it often causes a boom in the loudspeakers.

Each Input Channel will have an **On-Off** button of some kind. It may say ***Mute***, or it may have a ***Channel Number***.



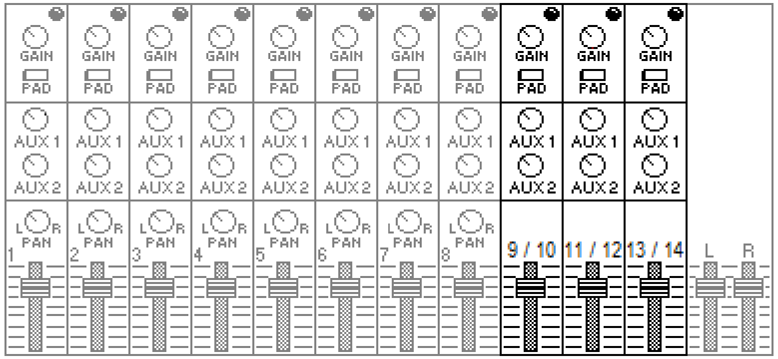
Some mixers offer ‘intermediary’ faders between the Input and the Output Channels.
These are called **Sub-groups** (Sub-mix) (Sub-master). They give you the chance to sum a number of Input Channels together into one, before continuing on to the main Left and Right Output faders.
By raising or lowering one of these sub-group faders you affect all the channels in that group.
Selector buttons on each channel let you assign a channel to a subgroup of choice, or directly to the main L-R instead.
Let’s say Channels 2,3,4,5 are all drum microphones. Press 3 (for example) on those four drum channels (2,3,4,5) to send them all to Subgroup fader 3.
Press L-R on Subgroup 3 to connect to the Output L-R Faders.
Now you can control all the drum channels by simply moving Subgroup 3 fader.
Harmony singers often have many microphones as well. Then there is the brass section, then the string section. Grouping sections of microphones lets you preserve the delicate balance you work so hard to attain.
We classify a Mixer by its number of channels and subgroups. Our example here has six input channels, four subgroups and two outputs (L-R). We call this a 6 on 4 on 2. If there were no subgroups, then it would be a 6 on 2.



A Fader is actually a Resistor. The further you pull the Fader down below zero, the more it resists.
If you look at a Fader you will see the **Zero** is ¾ of the way up.

Mixers are always designed to be working at *optimum electronic efficiency* when the whole system is operating at “Line level”.
The only way to get this is to have all Faders more-or-less sitting on the zeroes printed by the manufacturer, including the LR output Faders, and all volumes levelled using the Power Block on each channel.
The proof that a mixer is operating correctly, is the Output (L-R) level meters will be bouncing up and down touching the zero.
The first sign that an operator isn’t doing it right is an absence of electrical level on the Output level meters (which the manufacturer put there for a purpose). If you can’t keep all the Faders around zero (including the LR output Faders) because the Loudspeakers will be too loud, then adjust the gain on the Power Amplifier.

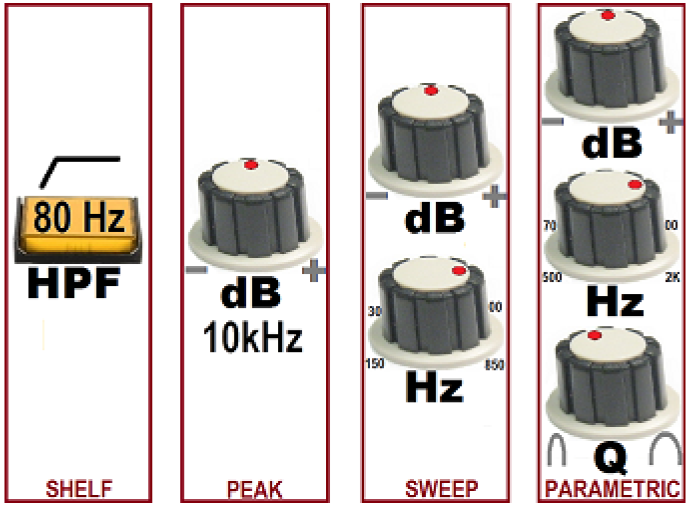
Fine adjustment: The Fader is meant for minor level adjustments.
Coarse adjustment: Once the fader moves too far from the zero, it is a job for the Gain control (up in the Power block).



Stereo Line-Input Channels

As well as the normal single channel (mono) inputs on a mixing console you may find faders with double numbers. These are two channel (stereo) inputs, and they will have two connectors (L-R) at the rear of each channel instead of one.
Our example shows 9/10, 11/12, 13/14. They are intended to connect devices that have stereo outputs (L-R) such as CD players, Keyboards, Laptops, Effect Units.
Radio Broadcasting mixers and D.J. mixers have mostly stereo input channels because they mainly deal with multiple stereo music players.

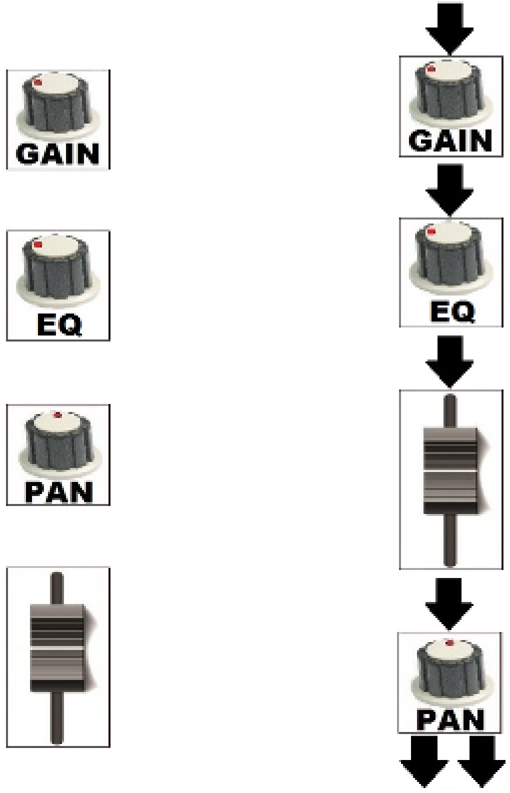
All controls (faders, knobs and buttons) on these types of channels are in stereo as well, and so they are affecting both the left and the right channel eg. 9 *and* 10.



Channel Equalisers / Filters

HPF: The High Pass Filter button cuts the bass frequencies. It is normally pushed on all microphone channels as it cuts out the rumble and bass noise below anything a human singer might be doing. It is called a Shelf Filter because it slides off the basses, the lower the bass frequency, the more the cut.
PEAK: This is a simple 'dial' filter that has a pre-set frequency (often a high frequency) which you can boost (+) to add freshness to a sound or cut (-) if the sound is too bright or contains hiss.
SWEEP: You use this Filter by boosting (+) just a little and then "sweeping" the frequency dial (Hz) back and forth. When you hear the problem area (which will be slightly exaggerated by your boost) then you can go back to your cut/boost (dB) dial and adjust it until things sound right.
PARAMETRIC: This Equaliser gives us control of all three filtering parameters. Once we "sweep" to find the critical frequency as mentioned in the Sweep filter, we can adjust how big an area of frequencies that will be cut or boosted (the Bandwidth or Q factor).
For example, you would select a narrow band if you wanted to lower a strong "Ess" sound in a voice so you don't filter away the rest of the freshness from the sound. You might use a medium band-width if you wanted to clean up a really boomy sound.

Mixers are designed with the Faders at the bottom of each module. This is only for your convenience, and it doesn't represent the true signal flow.

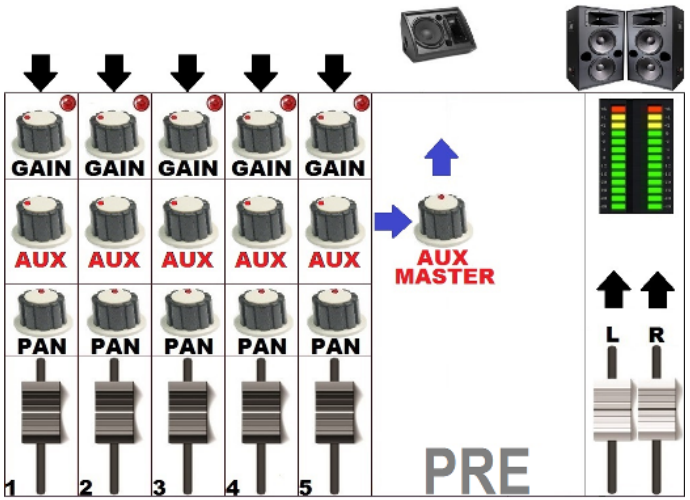


The column on the left shows how a mixer is laid out. The column on the right shows the actual signal flow.

The signal does not exit the channel at the Fader, but rather it splits in two after the Fader, and goes to the Pan (L-R).

Any knob or button that is in the signal path **before** the Fader is called **Pre-Fader** (or just ***Pre***). For example, the **Eq**ualiser is "Pre".
Any knob or button that receives the signal **after** the Fader is called **Post-Fader** (or just ***Post***). For example, the Pan is "Post".

Aux (Auxiliary) Sends



As well as sending our Input channels to the Left/Right Output Faders, we have the possibility to send additional signals to the Floor Monitors and Effect Units, etc.

We have Aux (auxiliary) Send busses on a mixer. They may be called Aux or Send.

At the end of each row of Aux Sends, there will be a Master volume (somewhere) so you can control the combined output volume of that row of Sends.

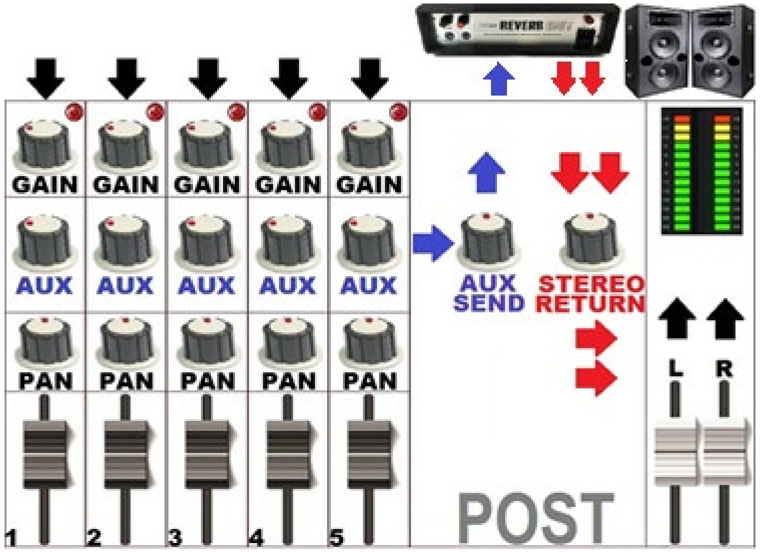
If a floor monitor gets too loud, and starts to squeal, then you can quickly turn back the Aux Master volume to stop it squealing (instead of turning down every Aux Send along the row and ruining your mix).

Normally the first few Aux busses (Aux 1, Aux 2) will be **Pre**-Fader. This is because the most common use for Aux Sends is to send signal back to the Floor Monitors (Fold-back).

A Send that is Pre-Fader will be unaffected by any movements made by the Channel Fader (*best scenario for floor monitor work*).

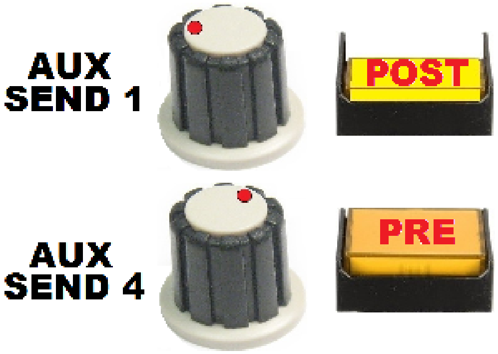
There is another kind of Aux Send buss, called **Post**-Fader. It will not be positioned on the mixer at the bottom, and will often be up with the Pre-sends. Somewhere it should be written “Post”.

This type of Aux Send receives the signal after it has passed through the Channel Fader, and if you lower the Fader then this Aux signal will fade out as well.
This suits Effect Units such as Reverb. If we fade out a Channel we want the Reverb to fade out as well (it sounds a bit scary still hearing the Reverb after the sound has gone).



When we use a ‘Post’ Aux Send to go to an Effect Unit, we will also need to receive the effected sound back into the Mixer.

Effect Units have typically got two outputs (L/R). Mixers often have some simple Stereo Returns meant for this purpose. Otherwise you need to use two spare Input Channels, or one Stereo Input Channel, to connect to the outputs of the Effect Unit.



Pre / Post Selector

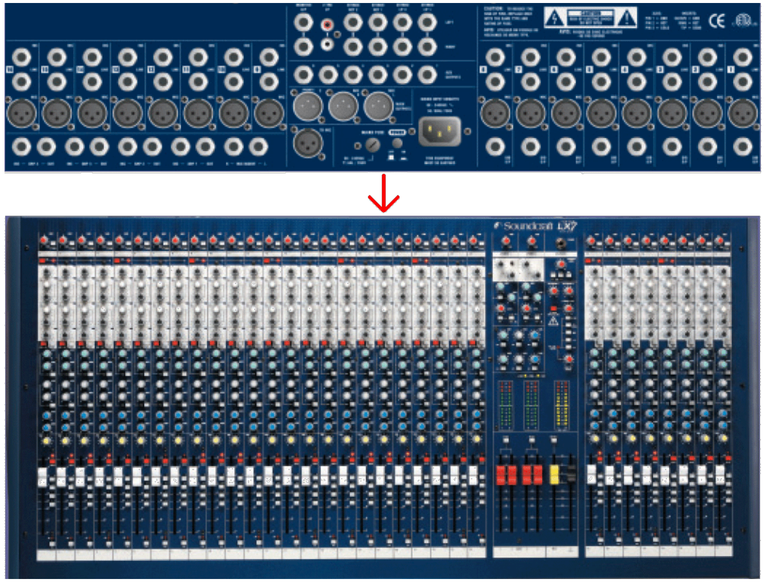
Some manufacturers give us a Pre/Post button on at least one Aux Send, so that we can choose to suit our needs.

If there is just a Post button, then you know it must normally be Pre.

If there is just a Pre-button, then you know it must normally be Post.

Auxiliary Sends will nearly always be in one central block. Sometimes it is not immediately obvious if they are Pre or Post Sends. If you turn one up, and send it to something (perhaps a floor monitor), and then lower the Channel Fader, the volume will fall if it is *Post*.

Watch your Back!

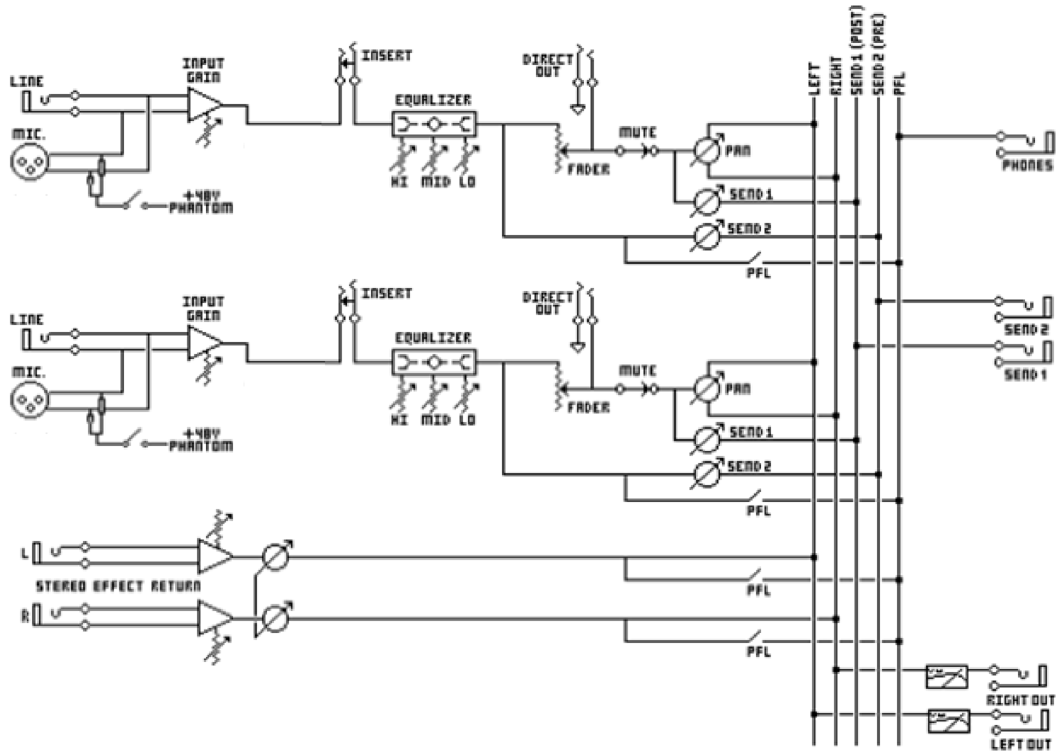


It is important that you familiarise yourself with the rear of an Audio Mixing Console.
Look at the connectors and their names, then find the matching names on the top.

Be sure you know clearly what they are all for. This may require looking at the manual, or experimenting. This helps you understand what is going where, and the particular names and labels the manufacturer is using (which varies).
It also teaches you options you have available to route things, particularly when you get those occasional complex jobs (they happen).

Knowing exactly what all the connectors do, and how you control them from above... will make you a powerful mixer operator. Sadly, only around 10% of people who operate mixers know their mixer that well, and they are not able to be there when the need arises.

Familiarise yourself with all points where the manufacturer has placed **Insert** connectors on the mixer. Insert points allow you to use Wye cables and insert Audio Effect units, particularly Compressors.
They can be used as Inputs if you are really needing some additional connections but have run out of input channels.



Sometimes manufacturers give the buttons and dials vague names, and then a terrible explanation in the manual so you're none the wiser. Then, just to confuse you more, they offer you some fancy "features" unique to that mixer, that they're convinced will help you, and they give them an even stranger description.
Whenever you are unsure about what things do, or where they go, it pays to look at the Block Diagram (usually in the back of the mixer manual).
It looks complicated at first, but just follow the lines with your finger, and you'll see where things sit in the signal path, and what names they use.
A good start is to search out the Line/Mic inputs and just follow the lines from there.

You will use a mixer with more wisdom and confidence when you know exactly where everything is located in the signal path.

OUTPUT CHANNELS

A mixer may offer a third ‘**MONO**’ fader at the output, occasionally it may be called ‘***SUM***’.
This is the sum of the Left plus the Right Channels (L+R) all mixed together eg. Mono.
This output is intended for anything you need.
It is the immediate choice for Subs (sub-woofer bins) which are in mono.
It is great for a single loudspeaker cabinet somewhere in the room (or another room).
You can use it for a personal monitor on a stand, or side-fill speakers, quite simply… whatever you like.

In live sound you don’t usually use stereo imaging (all Pans are kept in the centre, so the Left and Right channels both have the same content eg. Mono). So here the **Left fader** might serve Loudspeakers on the left side of the Auditorium, and the **Right fader** serve the right side, and the **Mono fader** serve Loudspeakers in a third area (children’s room, foyer, mezzanine floor etc).

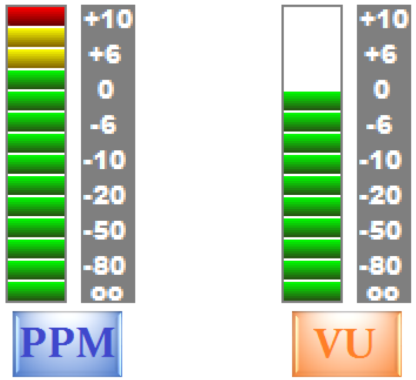
Level Meters

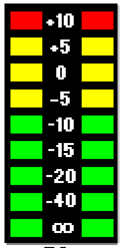
Level Meters on an audio mixer are special displays that have been slowed down for us to be able to see movement (sound waves move far too fast for human eyes).
As a consequence, when working with live sound, remember always that there will be electrical peaks that are a lot higher (happening a lot faster) than the slow Level Meter will have time to display.

Normally we don’t need to worry about occasional high peaks touching the top, because of the “headroom” (*the built-in safety buffer zone before things distort*) but if you are starting to get frequent flashing “Peak Lights” it is a sign that things are close to distorting. You may need to check… is the whole thing getting too loud, or is it one instrument that needs to be tamed down?

The *ideal level* is when the signal rises to the top quarter of the meter, only in loud moments.

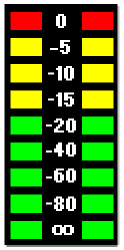
Both meters are receiving the same piece of music…





Vu

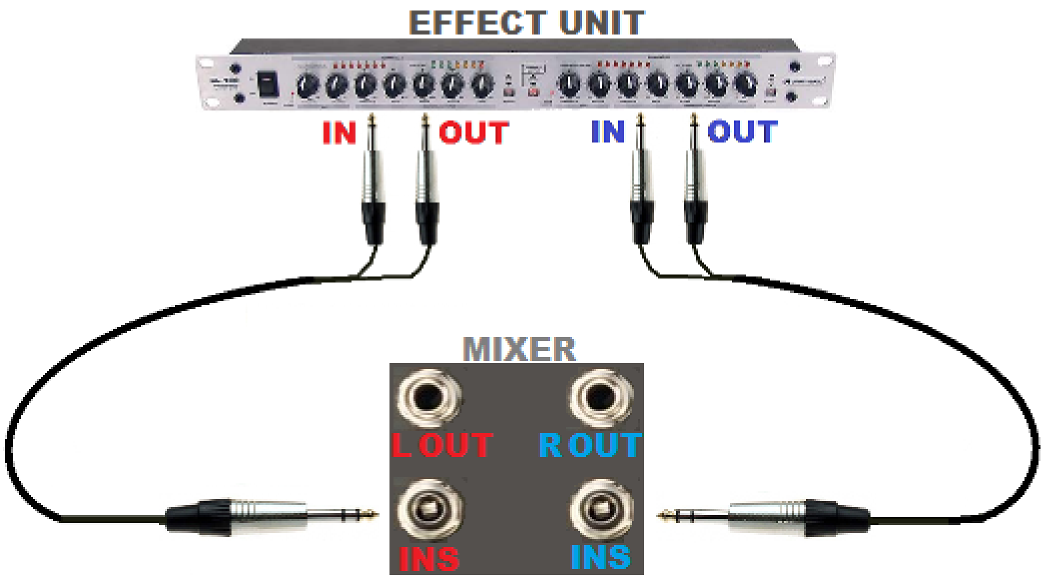
Vu Meter (***Volume Unit Meter***):
Vu meters are the normal Signal Level Meters you find on Mixers. Vu meters don't move particularly fast and are really only reading a more average version of your signal level. Because of this remember that the peaks are anything up to 30% higher than what you are seeing. Mixers know this, and there is a built-in headroom to allow for it. You just concentrate on having a healthy signal that splashes up near the top in its loudest moments.
What does VU mean? Unfortunately, the world never agreed on what voltage the zero on the meter should actually be. The Consumer Audio industry like it to be calibrated to ***–10 dBV 0.316V_{RMS} 0.894 V_{PP}****. "Some" of the Professional Audio industry like it to be calibrated to* ***+4 dBu 1.228V_{RMS} 3.472 V_{PP}****. America and Europe differ. So, we call it a V.U. (****Volume Unit)*** *meter because the actual unit of voltage at the zero marker depends on who made it.*



PPM

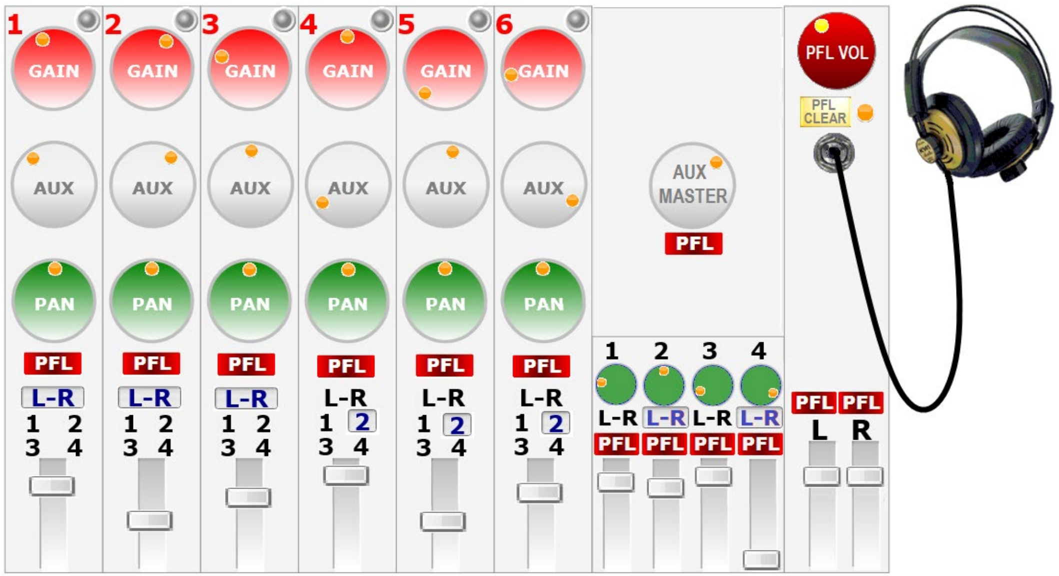
Peak Meter or **PPM** (***Peak Program Meter***):
As mentioned earlier Level Meters are slowed down for us to be able to see sound movement (sound waves move far too fast for the human eye).
The Vu meter which is only reading a bit more than the average voltage, is perfect for live work, however it isn't suitable for digital mixers or digital recorders who distort immediately if you go over their zero at the top.
A Peak Program Meter (PPM) lets you see the actual maximum electrical peaks (transients), though the meter has been slowed down. It is important to remember, if you find yourself working with a 'Peak meter', that what you are seeing is real, and any time the level reaches maximum you are extremely close to distortion.

Inserting Effect Units in the Stereo Output



The Main L-R Output connectors on a Mixer often have **Insert** points. These are for you to insert a **Compressor/Limiter** (with a wye cable). Used properly (lightly), it makes everything more intelligible, and improves the sound. If you have the budget you can insert a **Drive Rack** that does a multitude of wonderful things including stopping feedback squealing.

SOLOING



PFL (pre-fade listen) (piffle). This is for listening to selected channels discretely in your headphones, while the congregation continue to hear all channels in the Font-of-House Loudspeakers. Because PFL is pre-fader you can hear a channel even with it's fader down (and the congregation aren't hearing it at all). You can press more than one PFL button at a time and hear a combination. When you press PFL you may see the electrical level of your PFL appear on one of the Level Meters. Some mixers provide a PFL Reset or Cancel that turns off all PFL's that you might have pressed. You PFL an Aux Master Send to hear what a Floor Monitor is hearing. This is how you balance each monitor mix.

AFL (after-fade listen) is also heard using the headphones. Though most live mixing situations suit PFL, Digital Mixers often give you the option of PFL or AFL. Because AFL means you are hearing things 'after' the fader, if the fader is lowered then the volume in the AFL will lower. AFL is typically **SIP** (Solo-in-place) which means you will hear all of the panning, filtering, effects and levels for that channel 'in place'.

SOLO is a button that isn't used in live sound, as it is what we call '*destructive*'. When you press a SOLO, all other channels are cut off (muted). You can press many SOLO buttons at once. All channels that are not soloed are muted. You don't need the headphones as all changes come through the Front-of-House Loudspeakers. This is why we call it destructive, the regular sound is cut off and the congregation hear what you hear (just the soloed channels).
SAFE buttons (SOLO SAFE) sometimes exist that will let you 'lock out' the SOLO buttons so you can't bump them accidentally, although a mixer with Solo buttons is meant for the Recording Studio, and shouldn't even be used on Live Sound.
EXCLUSIVE Solos may be available. This type switch off (override) any previous Soloes that are pressed.

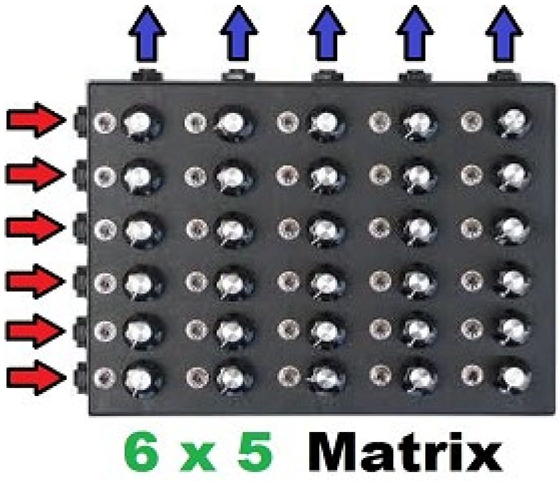

A **TALKBACK** button is for communicating with the Musicians through their Floor Monitors. There will be a microphone built-in to the mixer, or a socket for you to connect your own microphone. When you hold down the Talkback button your speech goes on to all Aux busses, which goes to the musician's Floor monitors (but not to the main L-R loudspeakers, so the congregation don't hear you communicating).
This is principally intended for Sound-checks but can be used discretely (in an emergency) during the meeting.
If the Mixer doesn't have a Talkback, you can connect a spare Microphone to a spare Input Channel and turn up all the Aux Sends (but leave the Channel Fader down). Now whenever you speak it will go the Floor Monitors but not to the Front-of-House.

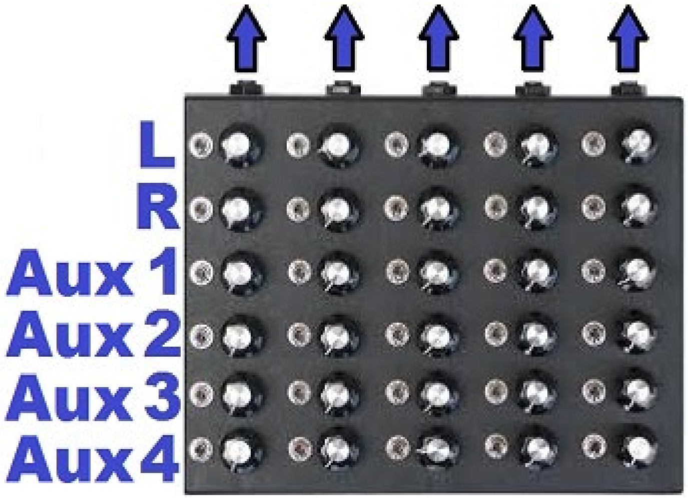
If you see a **MONO** button on the Monitoring Section of a mixer it is intended to help you check what will happen if your mix Left and Right sum together.
This is meant for Recording Studios where the mix is done in Stereo, and you want to check what happens in Mono (eg. if it will be Mono-Compatible).
Both the **SOLO** and the **MONO** buttons are for Recording Studio work, but some mixers are built for Live **or** Recording Studio, so you find these features.

The **DIRECT OUT** (D.O.) connectors are points that you can use to connect directly with the outside world (while continuing all your regular mixer activity with that channel). *Think of it as stealing a copy of the signal.* The Direct Outs are pre-fader outputs and so any fader movements made by the mixer operator won't affect what goes out of a D.O.
They can be used to split a channel and create a duplicate (use a phone cable to come out of the D.O. and connect it to the Line-in of a spare channel). This can be used to improve the sound of something (filter and effect the two channels differently, then mix them together).
The Direct Outs are what you use to connect to a Multitrack Interface for recording.
The Direct Outs are what you can use to connect to a separate specialised Floor Monitor Mixer (if you don't have a split stage-box).

GROUPING

Matrice (Matrix) Groups





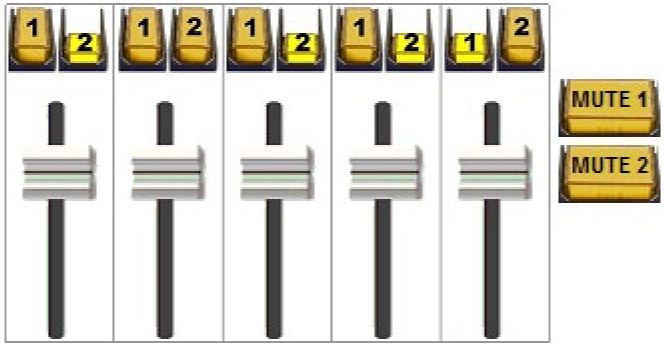
A Matrix is a place on the mixer where the sounds from different places all gather. The exact choices will depend on the manufacturer. Some matrix even permit you to add an external sound source (Ext). Your external sound won't be heard on the Mixer, just through the Matrix.

In this example matrix we have the ability to create five independent mixes (if we wish to). Each of these can be connected to a different destination.
One may go to additional speakers on the mezzanine floor, under the balcony, out in the foyer. One may go to an Induction loop in the church for hearing aids. One may go to side-fill loudspeakers on the stage. One may go to a live-recording television caravan parked outside etc. Each of these five mixes is unique.

If you want stereo, of course, you will need to use two channels.

With a Matrix there is sometimes the possibility to insert a Compressor, Equaliser, Delay etc.

Mute Groups



Some Mixers have Mute-Group buttons. This allows us to choose the channels we want to be involved in a group. If we press the Master **Mute 2** button, we will mute all those channels who had Group #2 pressed.
For example, if you put all the musicians on to a group, then you can mute them when they are not performing, eliminating musician noises and reducing feedback risk (less microphones turned on).

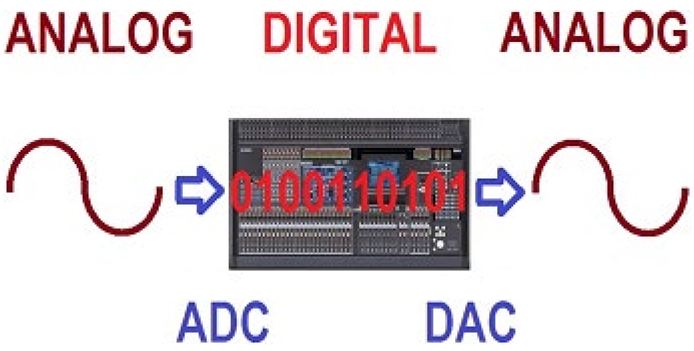
It is very flexible, and the members of the group are changeable at the press of a button.
A channel can even be a member of several groups.

Fader Groups

Some Mixers have Fader-Group buttons (often called VCA or DCA). This allows us to choose the channels we want to be involved in a group. The members of the group are changeable at the press of a button. A channel can be a member of several groups.
If we slide any of the faders in Group #2 we will change the volumes *by that amount* on all other members of that Group (which preserves the balance between them).
For example, if you have a lot of microphones on a drum-kit then, once the mix between the drum microphones is good, put them all on to a Fader group to preserve the balance, and now you just move any one of them to move the whole drum-kit. It is like a sub-group, but you don't have a separate sub-group fader.

By momentarily removing a Fader from a Group you can adjust it slightly, and then reinsert it in to the Group.


Church Audio
Digital Mixer Overview



A “Digital” Mixer receives the normal audio signal and immediately converts it to digital numbers, using an Analogue-to-Digital Converter (ADC).
The signal remains in this digital form as long as it is inside the mixer, then it converts back to analogue sound at the moment it exits the mixer, using a Digital to-Analogue Converter (DAC).

Working with digital numbers inside the Mixer allows us to do amazing things with the sound and offers many extra benefits and conveniences that regular Analogue Mixers can’t offer. The sound always remains pristine, even after endless manipulation.

Digital mixers offer you many ‘virtual’ busses and you can invent all the Aux Sends and complex signal paths that you want.

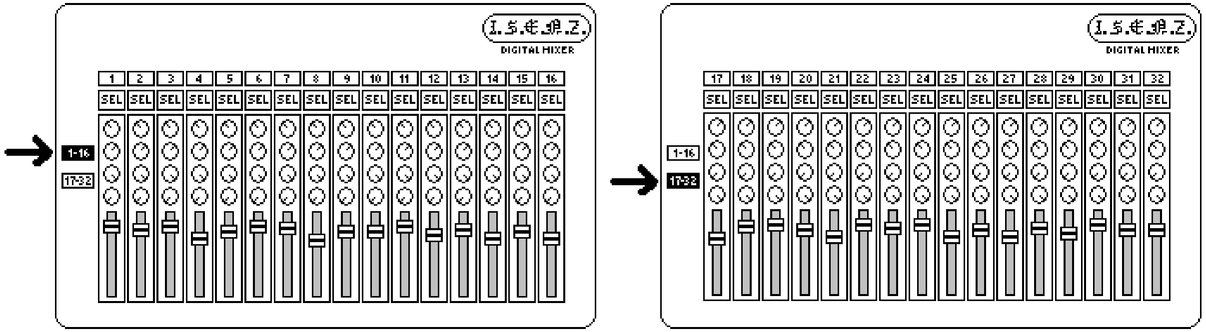
Professional Equalisation and Signal Processing, are available on every channel, so there are no excuses for the sound operator not making a decent sound.

USB Computer memory storage devices (which are digital) are often available for recording the Meeting. It’s simply digital copying what is at the L/R output.
WiFi Digital Mixers are sometimes Wi-Fi enabled, which gives us Remote Control options. With an ‘app’ placed on a Computer or Smart-phone you can use Wi-Fi to adjust the values on the Mixer while you walk around the church, hearing what the congregation are actually hearing, and adjusting there and then.

Note: Converting analogue to digital (A-D) and then back again (D-A) causes a slight delay (Lag). Normal (analogue) mixers don’t have this delay.
This digital ‘lag’ is too fast for us to hear, ***however*** if a digital and an analogue mixer are working off the same stage, and the sound from each is mixed together at any point, then you can hear phasing clashes. For this reason, always use “all digital” or “all analogue” mixers at a church event, not a mixture.

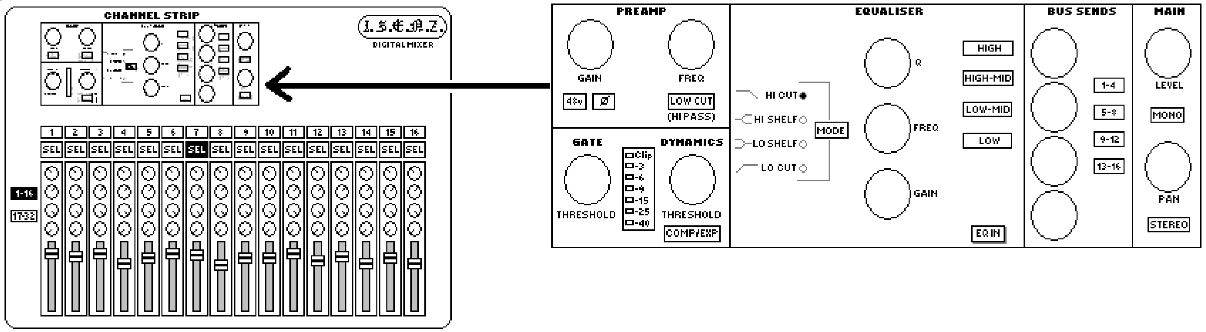
Digital mixers are laid out in a similar manner to Analogue mixers, however the dials, buttons and faders are really just 'controllers' that change a digital number as you move them. Here are the basics of Digital Mixer operations. Actual labels and positioning will differ with different models.

Input Channels:



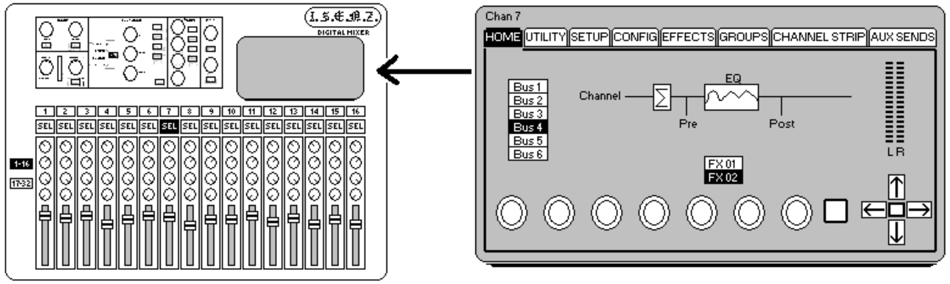
On an analogue mixer (normal mixer) every channel, knob, button and fader is visible. On a digital mixer not everything is visible, some things you need to "call up" to see. There may be 32 Input channels, but you can only see 16 at a time. By pressing a **Layer** (or **Page**) (or **Bank**) button you will see the others.
Our example above lets us access the first 16 (1-16) or the second 16 (17-32) bank of channels. This allows digital mixers to remain smaller than analogue mixers.

Channel Strip:



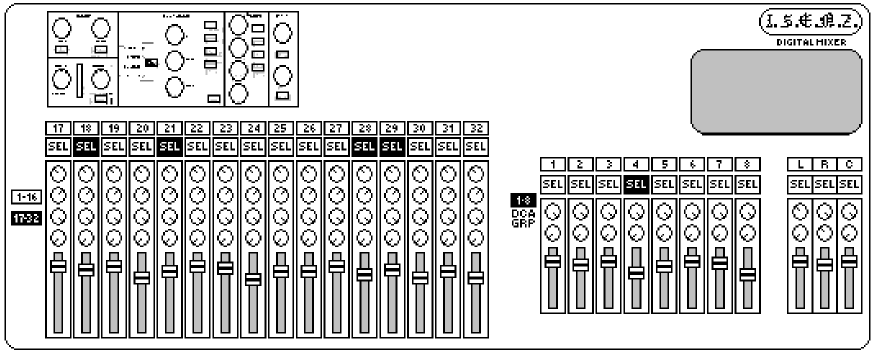
Normal procedure is to select (SEL) a channel, let's say "**7**". Now the mixer will display all settings for channel 7 on the Channel Strip display (Gain, Pad, Phase, and advanced Equalisers). Here you can make any adjustments you choose for Channel 7. Digital Channel Strips also offer Compressors, Limiters, Noise Gates. Channel Strips show the Aux Sends as well. Our example above shows 4 dials, but 16 Sends. Once again, things are in layers. Select a bank 1-4 or 5-8 or 9-12 or 13-16 (here you get 4 Aux Sends at a time to work with). Having things in layers and needing to call up a particular bank seems tedious at first, but it becomes second nature, and the benefits of Digital mixers definitely outweigh this singular idiosyncrasy.

Display / Screen:



Whenever you press one of the many select buttons (SEL) the “Screen” area will show you associated parameters, and settings related to that Channel. You can adjust parameters by touching the screen, using the cursor arrows, or turning the dials. The “Screen” normally changes to suit the “Select” button.
If you prefer to keep one particular screen continually in front of you, while you work, you can turn off the ‘**Screen-follows-Select**’ feature.

Assigning



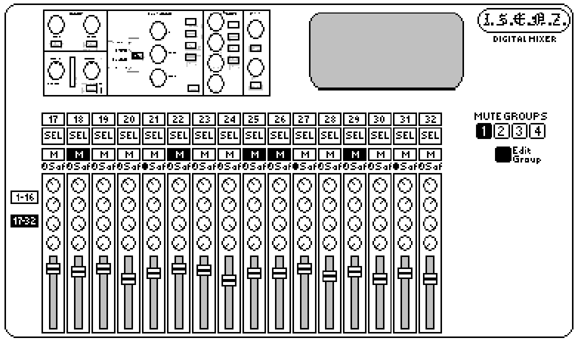
Like an Analogue Mixer, a Digital Mixer usually has a section of Input channels, then a section of Sub-group faders, and then the Output (L/R) faders.
Assigning to Master Output L-R: To send each Input Channel to the L-R or Centre (Mono mix) common methods are…

i) press a button labelled L-R on the Channel or Channel Strip, ***or …***

ii) press and hold the ‘Select’ button at the L-R Output faders, and then tap the ‘Select’ buttons on all the desired Channels to be connected to it.
Tap the Select button a second time to un-assign it.

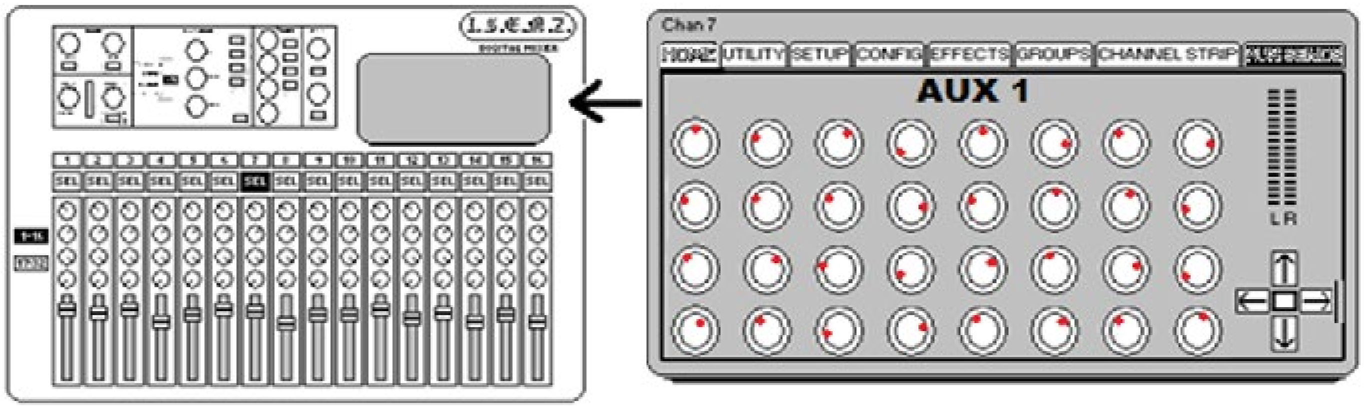
Assigning to Sub-Groups: When you wish to assign Input Channels to a Sub-group before going to the L-R, the method is commonly to press and hold one of the ‘Select’ buttons, let’s say Sub-group 4, and then tap the ‘Select’ buttons on the Input channels you want to be members of Sub-group 4.
Assigning methods will vary depending on the manufacturer, but the great thing is you that can set up the signal flow of the mixer to suit your needs, and undo any assignment by tapping things again. Assigning can sometimes be done, or edited, using the “Screen” Display.

Creating Mute Groups



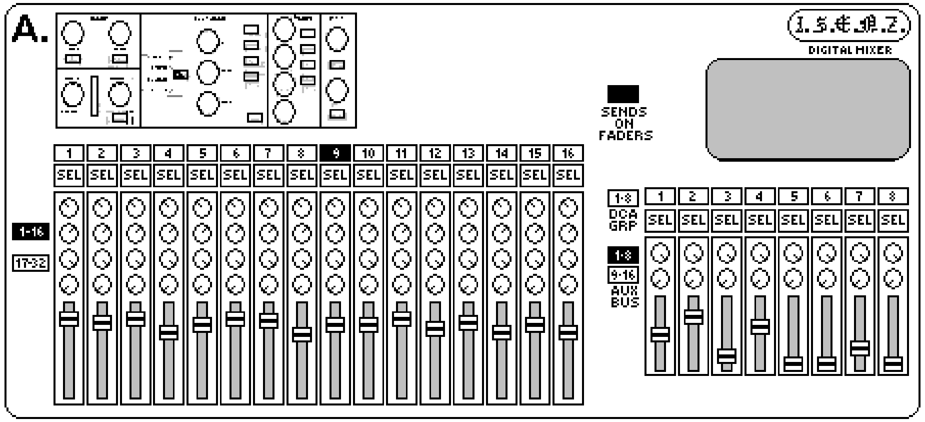
There will usually be some ‘Mute Group’ buttons. You may need to press an ‘Edit Group’ button, or turn off an ‘Edit Safe’ button, before you can make any changes. This is just a safeguard to protect the Groups from accidental changes.
Hold down one of the Master Mute Group buttons, let’s say #1, and still holding it down tap the Input Channel Mute buttons that you want to be part of this Mute group (tap any button a second time if you want to release it from the group). Channels may belong to more than one Mute Group, that’s fine.
Now you can mute all those Input Channels by simply pressing the Mute Group #1 button.

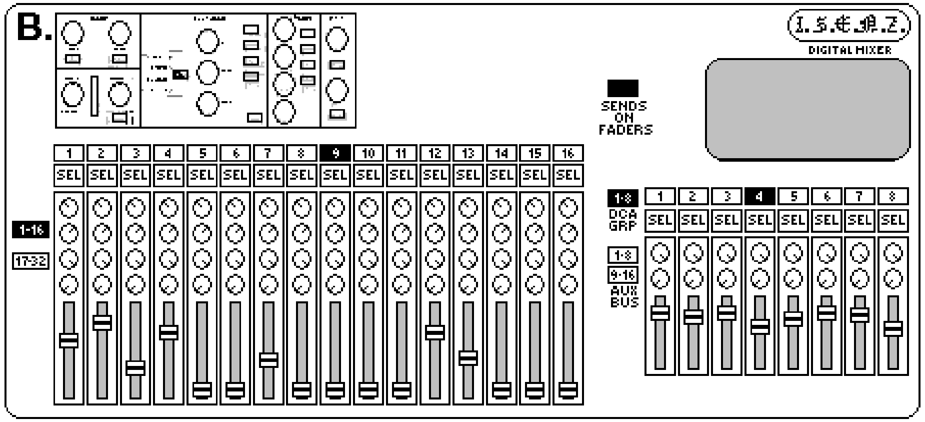
Aux Sends (Busses):



There may be 16 Aux Send Busses (or more) on a Digital mixer. The Display Screen can be used to display and adjust the Aux Sends, either by touch screen or using the screen cursors. This is where you can adjust a mix to a particular floor monitor.

Faders are used for adjusting many things (not just channel volumes):





Perspective A:
You want to look at all the Aux Sends belonging to one particular Input Channel (eg. to adjust what this channel is sending out to Floor Monitors):
Select the Input Channel, let's say #9. Now press the Aux Bus button 1-8 or 9-16 which will transfer the values of Aux Send 1-8 or 9-16 to the small group of faders. The fader positions have now moved to represent the Aux Sends, now you can adjust as you wish.

Perspective B:
You want to look at one particular Aux Send and see what each Input Channel is sending to it (eg. make a mix for a Floor Monitor):
Press the Aux Bus button 1-8 or 9-16, and select the Aux Send you want to look at. We'll say Aux Send #4.
Press the button that gets you '*Sends-on-Faders*'. Now the values of all the large Input Channel fader block will change, and the fader positions will move to represent the what each channel is sending to Aux Bus 4, which you can adjust as you wish.
Press the '*Sends-on-Faders*' button again and the Input Channel faders will return to their normal Sub-group values.

*All of this is possible because the **Faders** are really just **digital Controllers**. No signal actually passes through them.*

Snapshots/Scenes:

Because Digital Mixers only use Digital Controllers (rather than real knobs and faders) they just contain a number value, which can easily be memorised (stored in a memory). This allows us to take a Snapshot (Photo) (Scene) of the mixer settings and layout (the current values of all faders, buttons, dials, and signal flow configuration) in that moment. The Snapshot may ask for a name (call it what you like) and it will be saved into memory.
You can recall that Snapshot (Scene) at any time; and all controls will return exactly as they were when you took the Snapshot.
Note: To help you just recall "some" channels, any channels set to 'Safe' are not affected by recalling a Scene and will remain as they were.
How a Fader responds to a memory recall will depend on the type of fader it is. If the fader or dial is motorised (imagine a tiny motor inside) then it will physically jump to the memorised value, otherwise there will be a row of tiny lights beside the fader that will show the internal value that as is now.

CREDITS

This material is offered freely to the Christian Churches; downloadable at Pietango.com

Text: *Original, by the Author, a Christian Recording Engineer.*
Images: *Designed by the Author. Some photographs were sourced from the Internet, then re-worked.*

Ever since the creation of the world, God's invisible attributes and divine nature have been evident. They are clearly understood through his workmanship, and all the wonderful things that he has made. Therefore, those who fail to believe and trust in him are without excuse, or defence. **Romans 1:20**

All of us have sinned and fallen short of God's glory, but God treats us much better than we deserve.
Because of Christ Jesus, he freely accepts us and sets us free from our sins. God sent Christ to be our sacrifice. Christ offered his life's blood, so that by faith in him we could come to God. **Romans 3:23**

If you declare with your mouth, "Jesus is lord," and believe in your heart that God raised him from the dead, you will be saved. For it is with your heart that you believe and are justified, and it is with your mouth that you profess your faith and are saved. **Romans 10:9**

For the Scripture (*Isaiah 28:16*) says, "Whoever believes in Him will not be disappointed." **Romans 10:11**

These things have been written so that you may believe that Jesus is the Christ, the son of God; and that by believing, and relying on him, you may have new life in his name. **John 20:31**