A Basic Guide to Sound



Contents

Page 3 – What is Sound?

Page 5 – Cables and Connectors

Page 12 – Microphones

Page 21 – Amplifiers

Page 22 – Speakers

Page 24 – Crossovers

Page 27 – Ohms

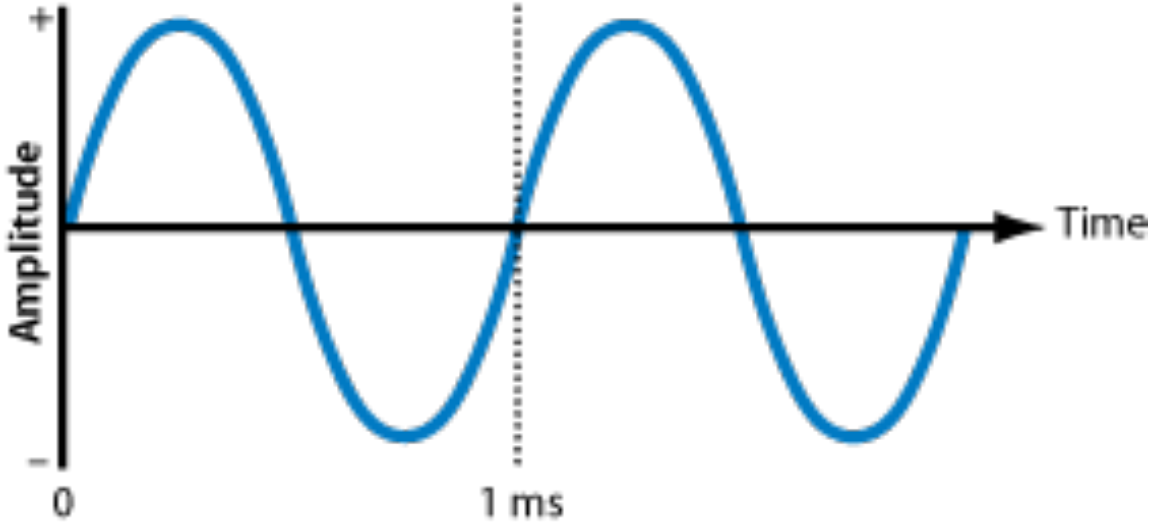
Page 31 – Mixing Desks

What is Sound?

All sounds are vibrations travelling through the air as sound waves. As sound travels, it reflects off objects in its path, creating further disturbances in the surrounding air. When these changes in air pressure vibrate your eardrum, nerve signals are sent to your brain and are interpreted as sound.

Fundamentals of a Sound Wave

The simplest kind of sound wave is a sine wave.



A sine wave clearly demonstrates the three fundamental characteristics of a sound wave: frequency, amplitude, and phase.

Frequency

Frequency is the rate, or number of times per second, that a sound wave cycles from positive to negative to positive again. Frequency is measured in cycles per second, or hertz (Hz). Humans have a range of hearing from 20 Hz (low) to 20,000 Hz (high). Frequencies beyond this range exist, but they are inaudible to humans.

Amplitude

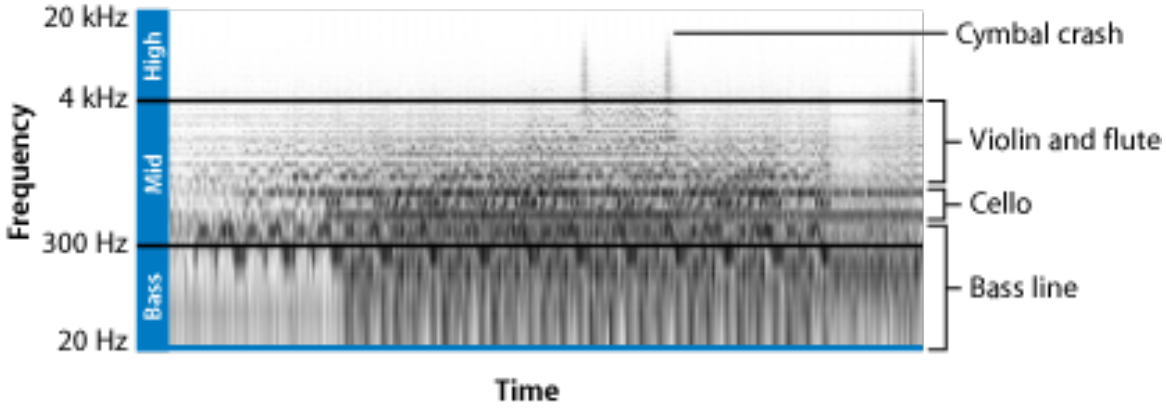
Amplitude (or intensity) refers to the strength of a sound wave, which the human ear interprets as volume or loudness.

Phase

Phase compares the timing between two similar sound waves.

Frequency Spectrum of Sounds

With the exception of pure sine waves, sounds are made up of many different frequency components vibrating at the same time.
Sounds contain energy in different frequency ranges, or bands. If a sound has a lot of low-frequency energy, it has a lot of bass.
High-frequency energy beyond the midrange is called treble.



Cables & Connections

There are two main types of sound cable: signal cable and speaker cable.

Speaker Cable

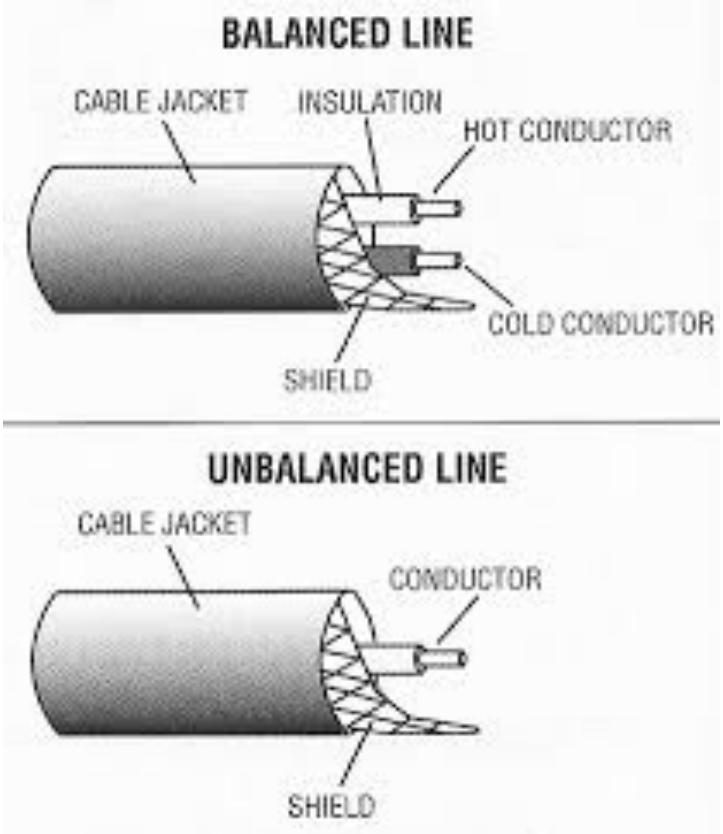
Speaker cable provides a path of low resistance between the amplifier and the loudspeaker.
Speaker cable is made up by:-
2 core cable: 1 core is called the positive (+) The other is called the negative (-)
Speaker cable is the cable used between amplifiers and speakers.



Signal Cable

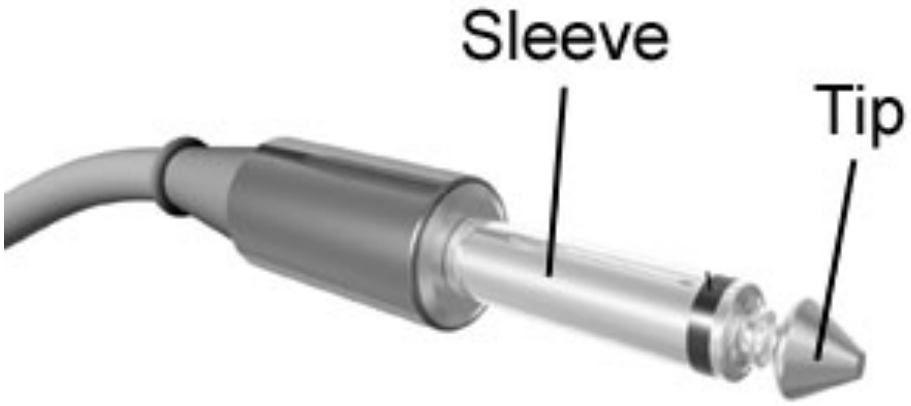
There are two types of signal cable: balanced and unbalanced.
All signal cable should be screened because the screen removes electrostatic interference.

Signal cable is made up by:-
2 core unbalanced cable or a 3 core balanced cable.
Core number 1 is the signal send known as + or hot.
Core number 2 is the signal return known as - or cold.
In a balanced cable core 3 is the screen.
In an unbalanced cable core 2 also acts as the screen.
An unbalance cable is synonymous with noise and interference, which is why its not used now in professional sound systems. It is still used with guitars and keyboards accompanied with a DI box, but never with microphones.



Connectors

Unbalanced Jack

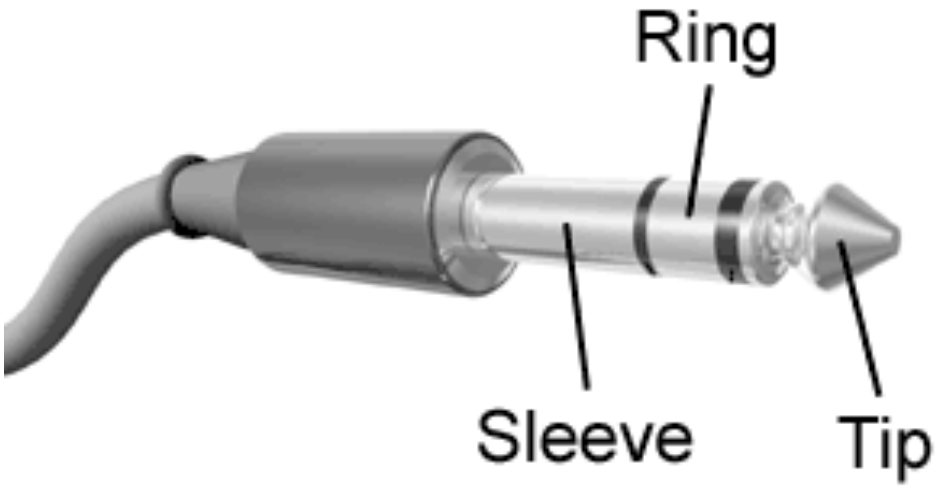


Tip = Positive Sleeve = Ground

AKA, Unbalanced Jack, TS or Mono Jack

TS is the abbreviation for Tip Sleeve and refers to a specific type of 1/4" connector that is set up for two-conductor unbalanced operation. An insulator separates the tip and sleeve. The tip is generally considered the "hot," or the carrier of the signal, while the sleeve is where the ground or shield is connected. TS cables are best known as guitar or line-level instrument cables.

Balanced Jack

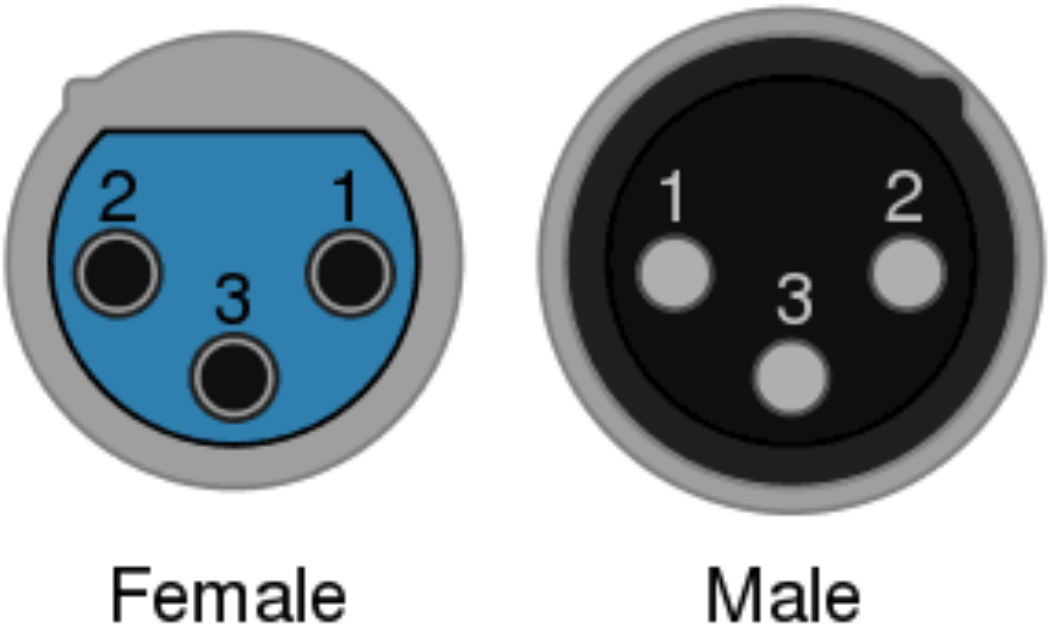


Sleeve = Ground Tip = Positive Ring = Negative

AKA, Balanced Jack, TRS or Stereo Jack

TRS is the abbreviation for “Tip Ring Sleeve.” A TRS plug can be found at the end of most headphone leads. It looks like a standard 1/4" plug with an extra “ring” on its shaft. The three sections of the shaft are called the Tip, Ring, and Sleeve. TRS connectors are used wherever you need to have two conductors plus a ground (shield) in one plug. A common use is to connect balanced equipment.

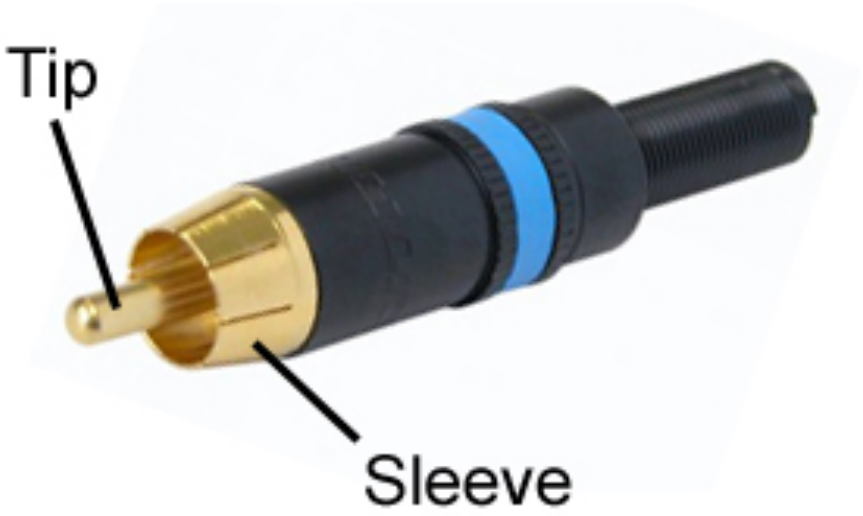
XLR



Pin 1 = Ground Pin 2 = Positive Pin 3 = Negative

XLR is the trademarked name for circular 3-pin connectors developed by Cannon. XLR has since evolved into a generic industry term, and many manufacturers now make this style connector, in which there are positive, negative, and ground connections. In audio, XLR connectors are normally used for transmitting balanced mic and line level signals to mixers or audio to amplifiers.

Phono or RCA



Tip = Positive Sleeve = Ground

RCA connectors are a common name for phono connectors like the ones used to connect most consumer stereo equipment. These were so associated with the RCA Corporation in the early 1900s that they became known as the RCA connector. Some mixers have tape or CD inputs and outputs with RCA connectors.

Speakon



Contains positive and negative poles

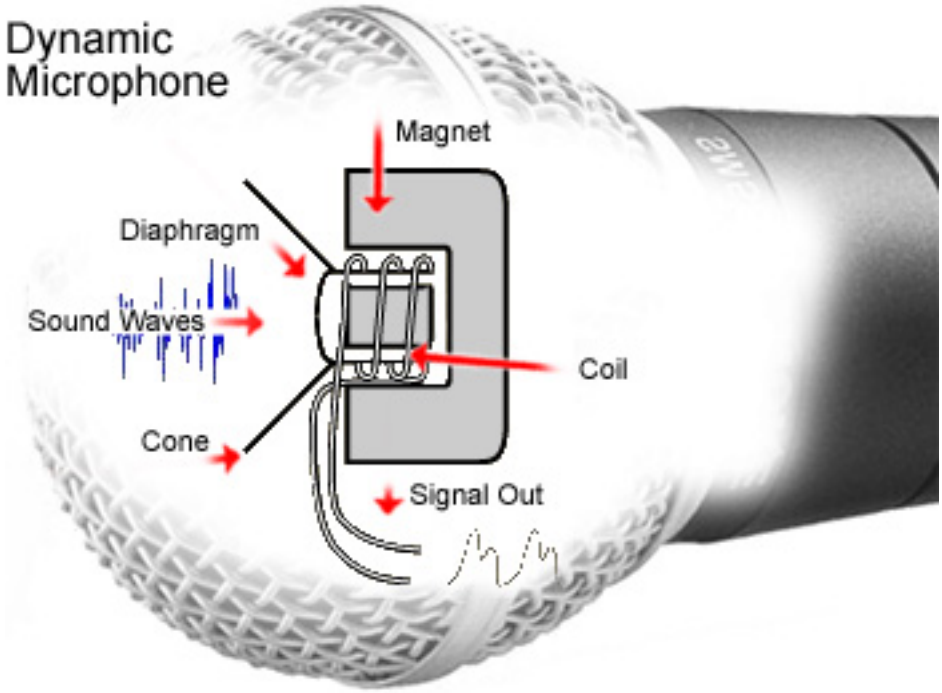
Speakon connectors are used to connect speaker cable to loudspeakers and amplifiers. They are a robust plug that locks into a speakon socket by ¼ of a turn to the right.

Microphones

All microphones have one thing in common:- Microphones convert a sound wave into an electrical signal.
To perform this task microphones use a thin membrane, known as a diaphragm, which mimics the function of the human ear
When sound waves strike a microphone's diaphragm, they cause it to move within an electro-magnetic field, which in turn creates a variance in an electrical current. This signal is then transmitted to output devices, such as amplifiers and mixers, which reproduce the original sound wave and reinforce it

Dynamic Microphone



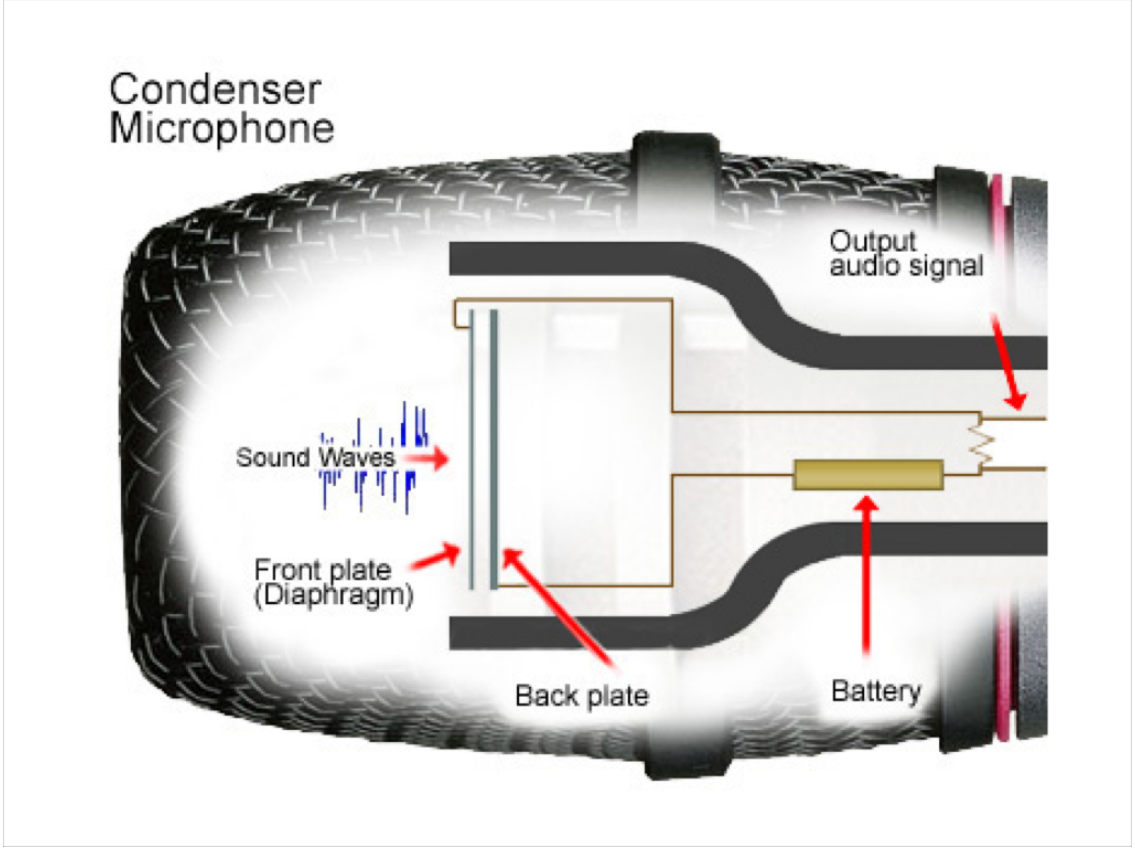


In dynamic microphones a physical cone acts like a lens to concentrate the incoming sound waves. The concentrated energy of these sound waves moves the cone and its attached coil of wire back and forth inside a magnetic field. The magnetic field induces electricity to flow through the wire to produce an electrical signal, which is the microphone's output.

Dynamic microphones are well known for their versatility and their durability, but are not the best choice for reproducing high and low frequencies; dynamic microphones require significant energy to move the coil of wire and therefore lose definition at the extremes.

Condenser Microphone



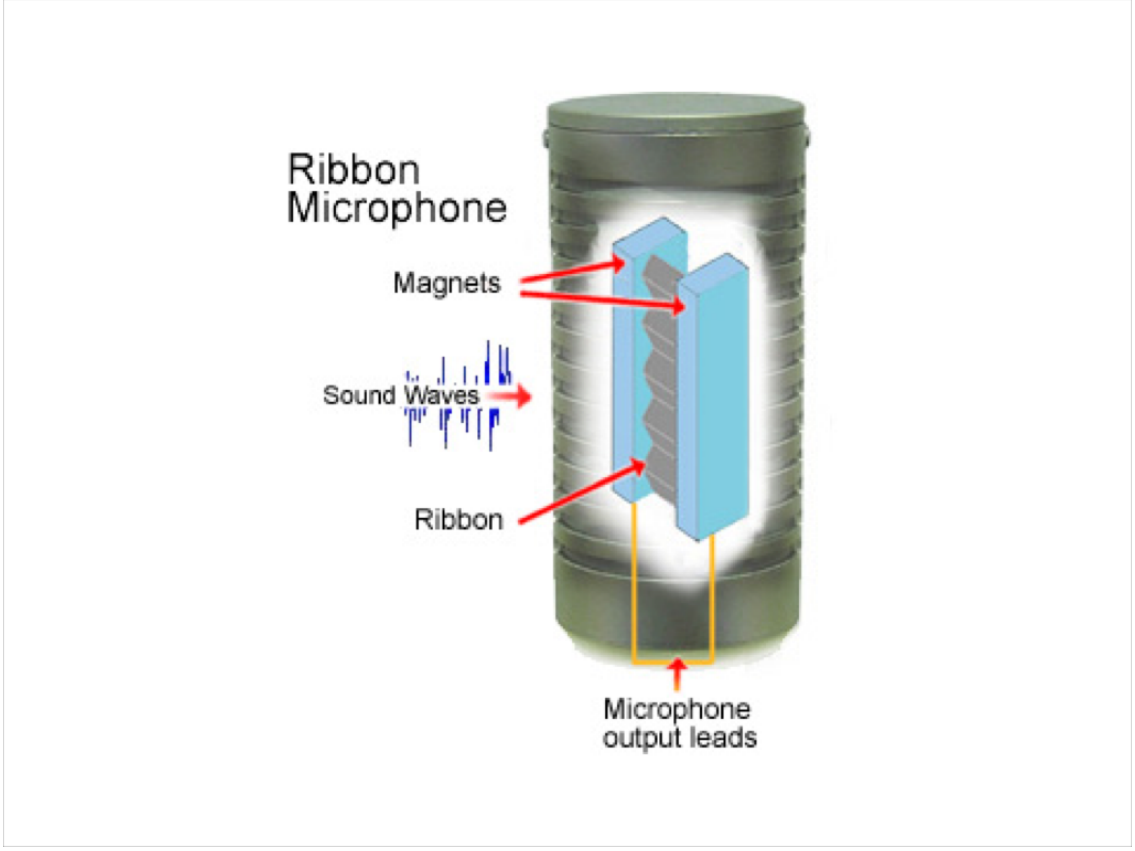


Rather than a vibrating wire coil, condenser microphones have a thin diaphragm and solid back plate which make up an electronic component known as a capacitor. As the diaphragm vibrates, the distance from the back plate to the diaphragm varies accordingly. This is known as fluctuating capacitance. It's the fluctuation that produces an electrical current, resulting in the signal output.

The electrical component needs to be powered. Most mixers have what we call phantom power, which sends 48V DC down the XLR to power the capacitor.

Ribbon Microphone



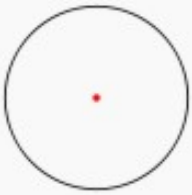
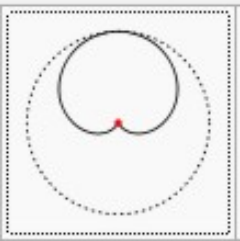
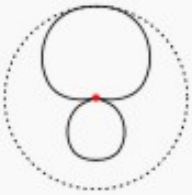
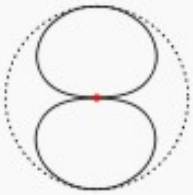
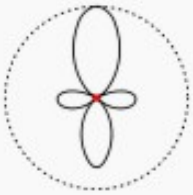


Ribbon microphones consist of a thin strip of metallic foil suspended in front of a magnetic plate. Sound waves cause the foil to vibrate, producing fluctuations in the electrical current, creating the audio signal.

This extremely sensitive configuration picks up a wide range of frequencies and produces an extraordinarily rich representation of the original sound. The trade off for this sensitivity is the delicacy of the mechanism. These ribbon microphones are also sensitive to physical impacts and therefore are not considered portable and can be easily damaged by power overdrive.

All microphones have what we call polar patterns. Basically they are designed to pick up sound from certain points on the microphone.

Here are some of the most common polar patterns:

Common polar patterns for microphones (microphone facing up in diagram)				
				
Omnidirectional	Cardioid	Hypercardioid	Bi-directional	Shotgun

Cardioid



Supercardioid



Omnidirectional



Figure of 8 - bidirectional



DI Box



A DI unit, DI box, direct box, or simply DI (variously claimed to stand for direct input, direct injection or direct interface), is a device typically used to connect a high - impedance, line level, unbalanced output signal to a low - impedance microphone level balanced input, usually via XLR connector.

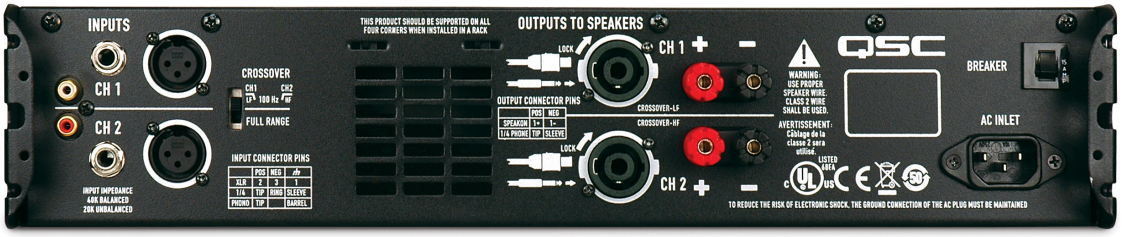
Amplifiers

Amplifiers take a tiny alternating electrical signal that represents the moment-to-moment variations of musical frequencies and their amplitudes (volume levels), and increases their strength many times so they're powerful enough to drive the cones and domes of speakers back and forth to generate air pressure variations (waves), which replicate the original sound waves.

From the front, amplifiers usually have a power switch and channel volume controls. In the picture below this amplifier has two channels.



The rear side has inputs and outputs. Usually a professional amplifier only offers XLR inputs. This amplifier offers XLR, Jack and Phono. Its outputs are usually Speakon or terminals allowing a bare ended speaker cable to be securely attached to the amplifier. This amplifier allows Jack outputs also.



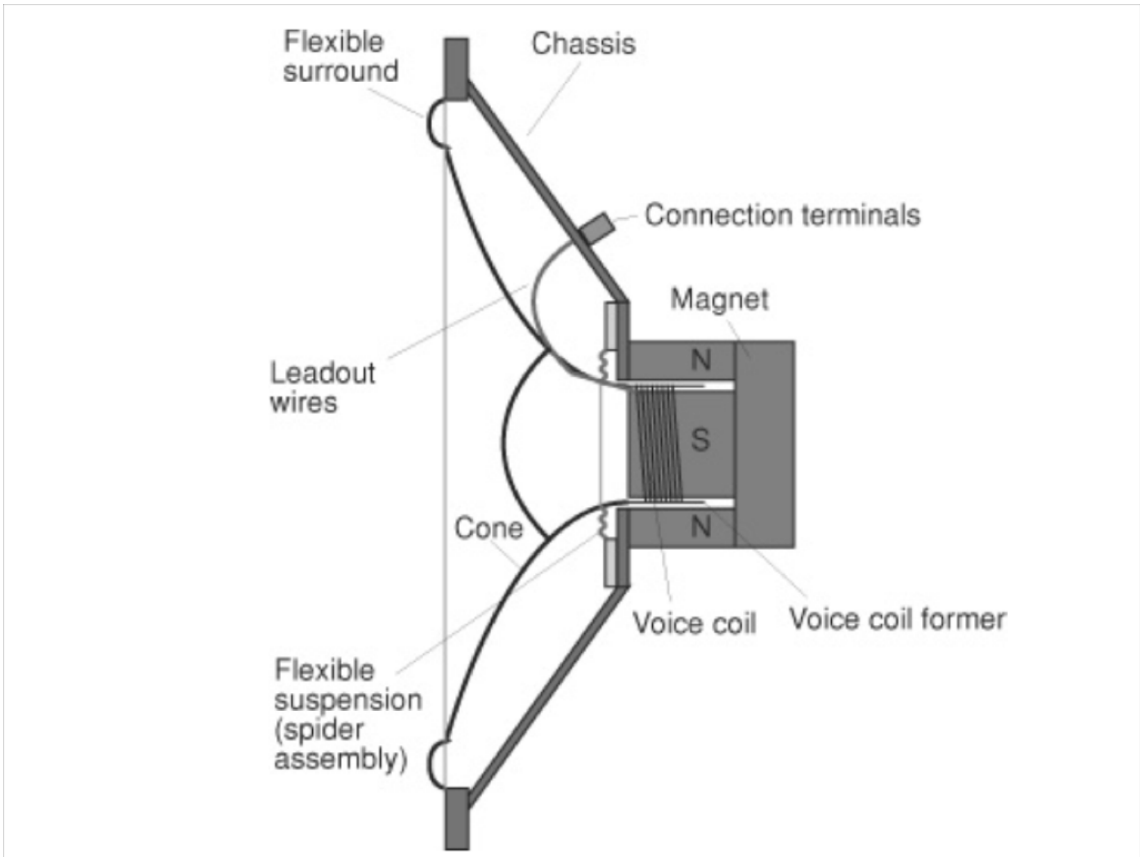
The more money you spend on an amplifier then the more controls and features it has. The amplifier below has various controls and costs into the thousands of pounds.



Speakers

A chassis loudspeaker without a box or cabinet is called a driver.

The driver is made up of a stiff cone of doped paper or synthetic material suspended in a rigid cage or chassis by means of a flexible surround.



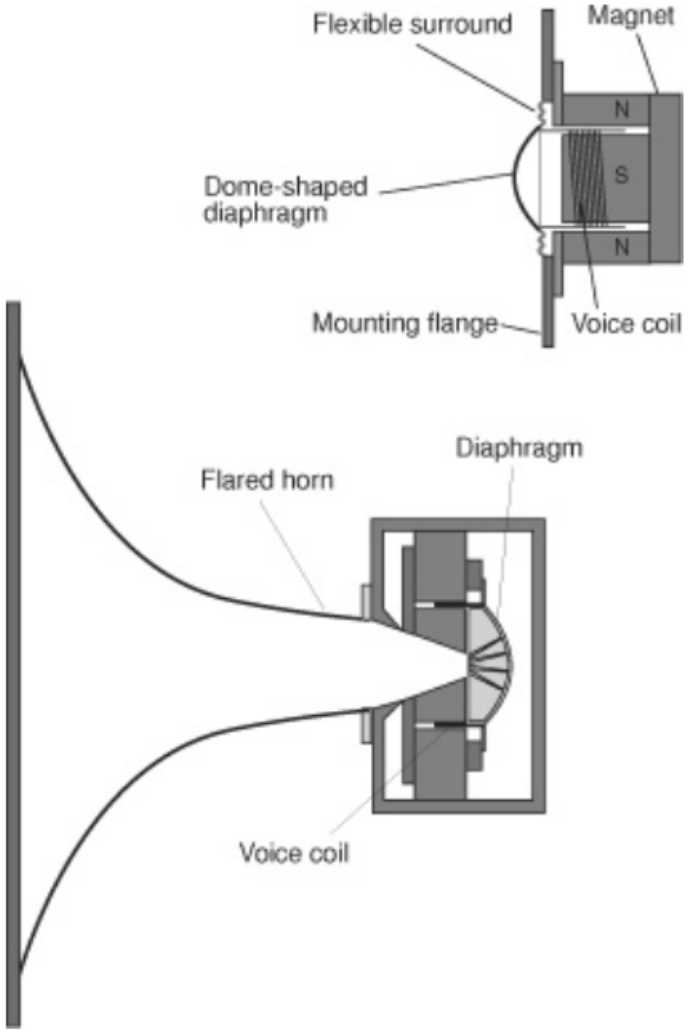
The narrow end of the cone is fixed onto a parallel-sided tube, onto which is wound a coil of thin wire, and the coil is positioned in a slot between the poles of a powerful magnet. When current passes through the coil a force is set up between it and the magnet, causing the cone of the driver to move either backwards or forwards on the polarity of the current. If the current passed through is an audio signal then sound is produced from the cone.
The driver is limited to how far the coil can move within the gap and the physical limitations of the cone within its chassis.

The harder you push the driver, the more it moves. Push it too far and it will physically tear apart causing to driver to stop working – also known as blowing the speaker.
To overcome working drivers too hard we introduce more drivers designed to handle different frequencies.

For example bass (low frequencies) drivers have to have a relatively large diameter.
The larger the cone then the more rigid it has to be as not to bend or deform under load.

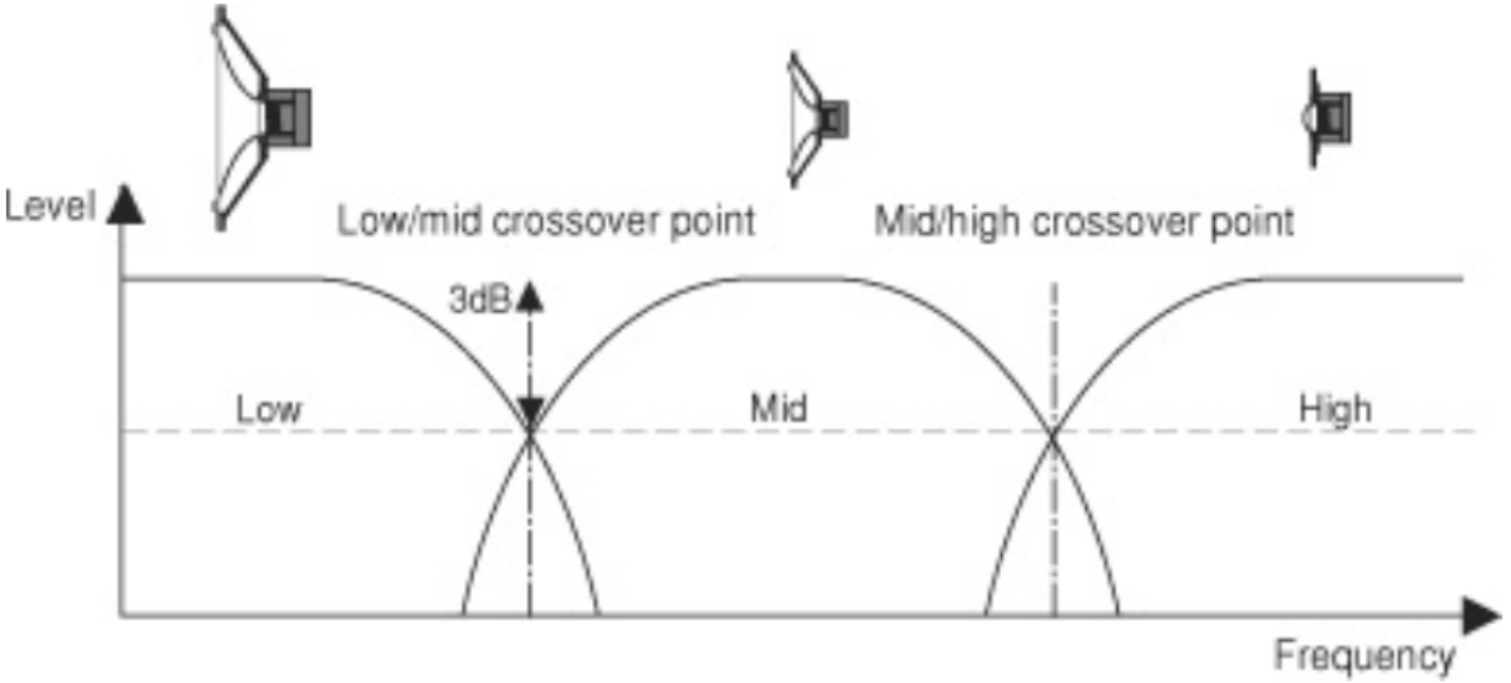
Mid-range frequency drivers have a smaller diameter than bass drivers because they don't handle as much stress.

Treble (high frequencies) don't need the rigid structure of a bass driver. They are called tweeters and are made up slightly differently. In larger sound systems the tweeters have a large horn to throw the sound further.



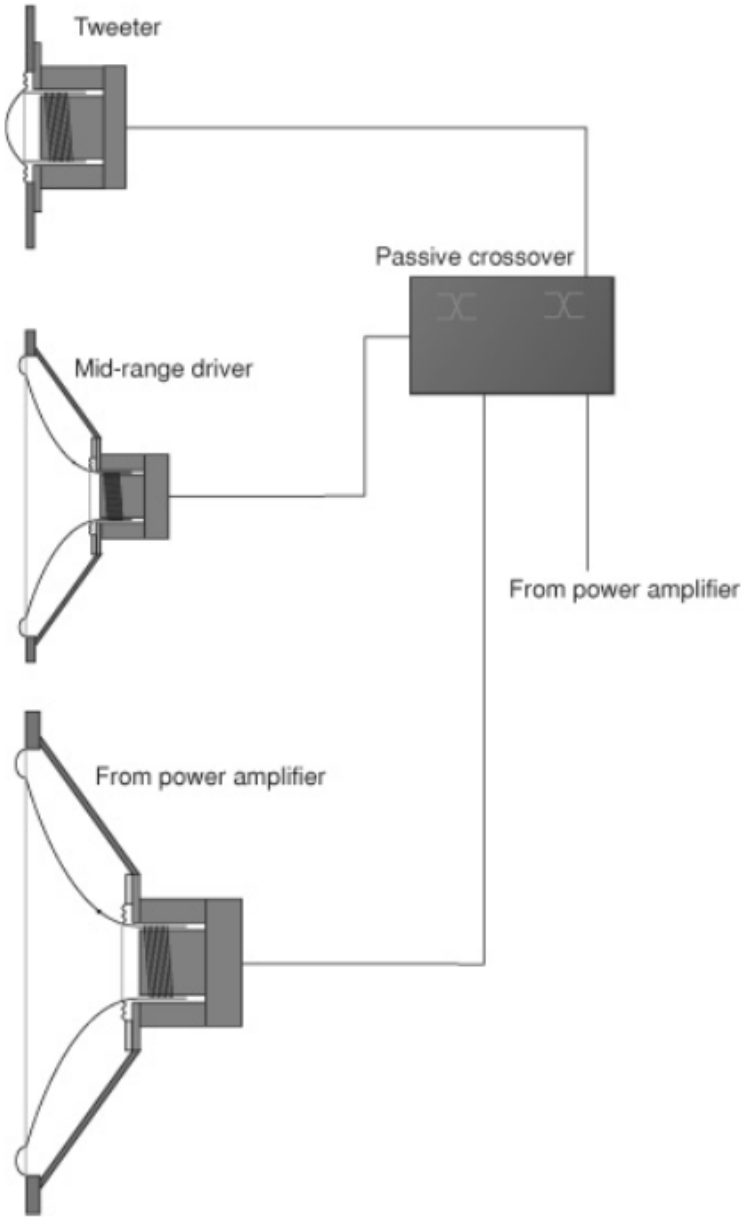
Crossovers

In order for the correct frequencies to be sent to each speaker the signal from the mixing desk must go to a crossover first.



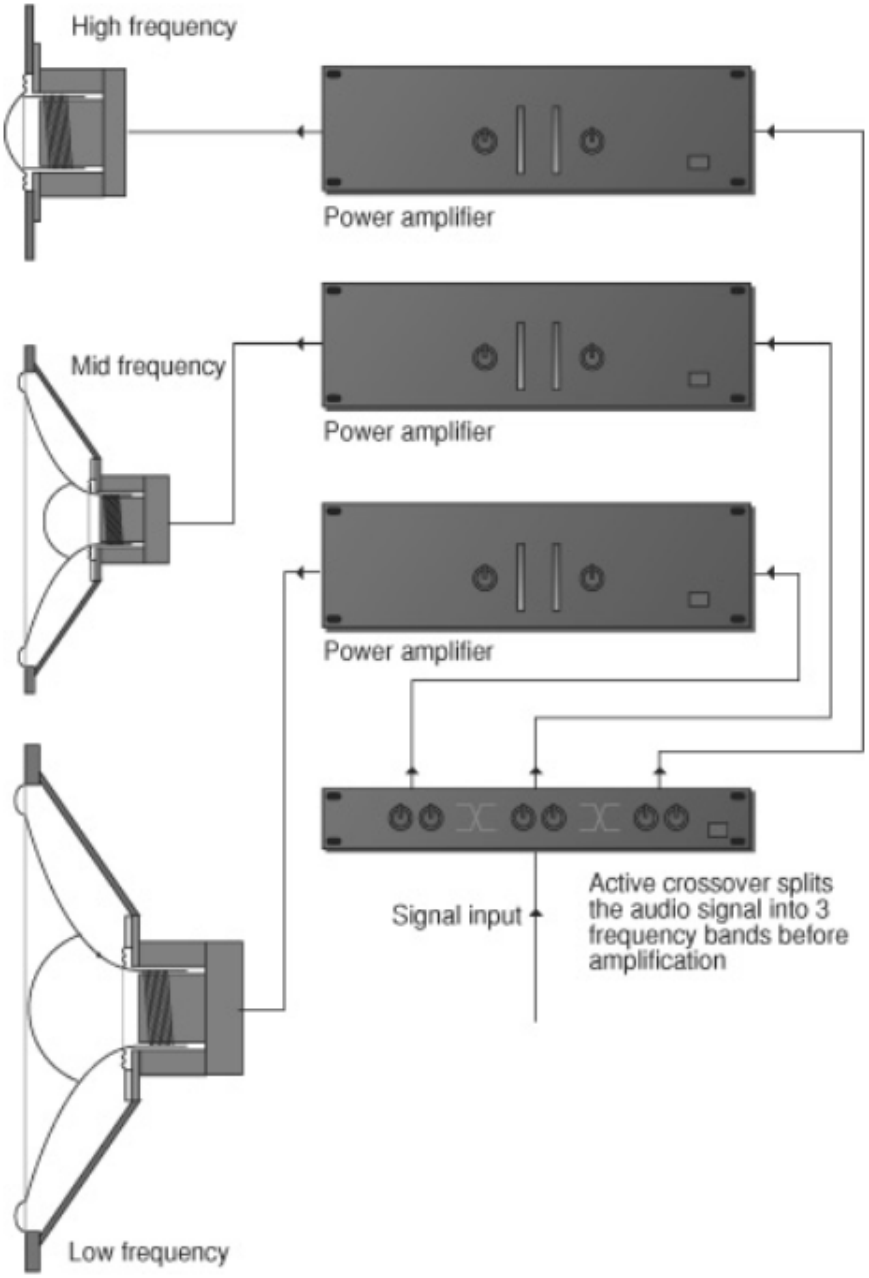
In a 3-way system, comprising of bass, mid and tweeter units, the bass speaker would be fed via a low-pass filter, which only allows through frequencies below a certain limit.
This ensures that the bass speaker never has to deal with frequencies higher than those it was designed to reproduce.
A mid-range speaker has both upper and lower limits of safe operation, so it has to be fed via both high and low pass filters to ensure it receives only mid-range frequencies. Frequencies that are too low will cause damage and distortion, whereas high frequencies will result in beaming.
High frequency drivers are fed via a high-pass filter to ensure that they receive frequencies only above a certain limit, so it's important that you select bass, mid and high frequency drivers that overlap slightly in terms of the ranges they can cover to ensure a seamless cross-over transition.

Passive Crossovers



Passive crossovers are designed for low power and tend to be found inside speaker enclosures that contain two or more drivers.

Active Crossover



Active crossovers are the typical set up in PA arrangements. They are found after the mixer but before the amplifiers.

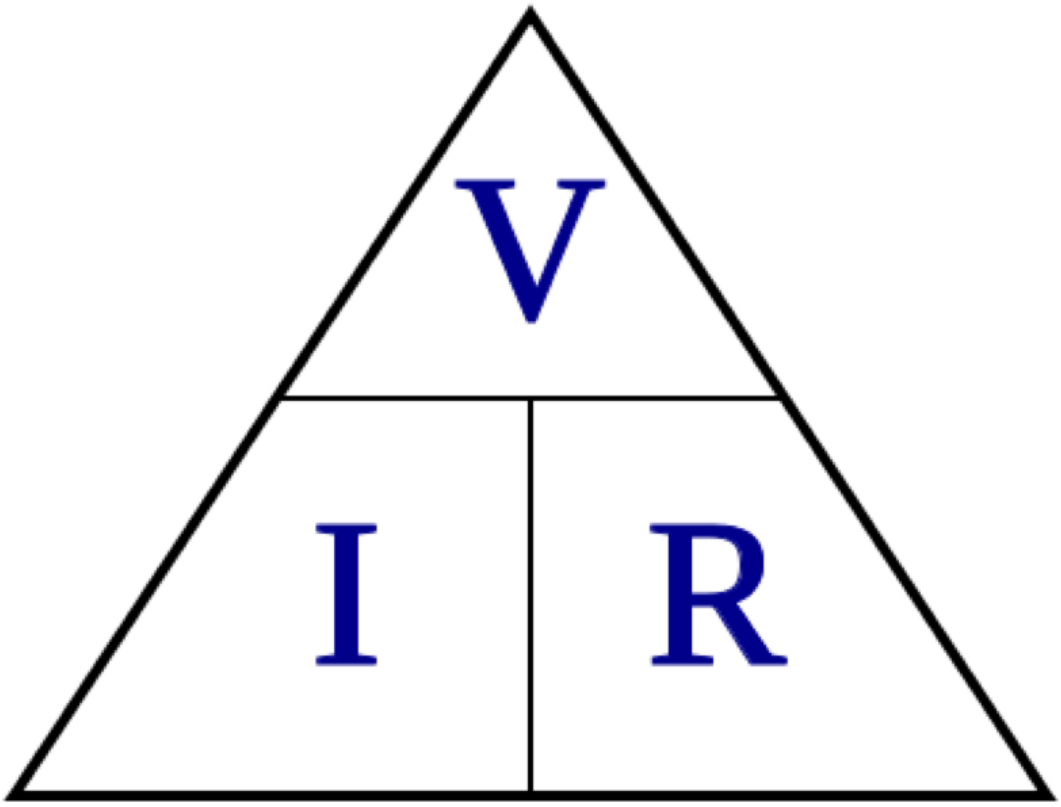
Ohms

Ohms are a measure of resistance. You don't think of it, but substances conduct or resist electricity on a scale. Rubber isn't very conducive, copper is. This is why some premium cable is made out of silver - silver & other expensive materials - they conduct electricity better than others.

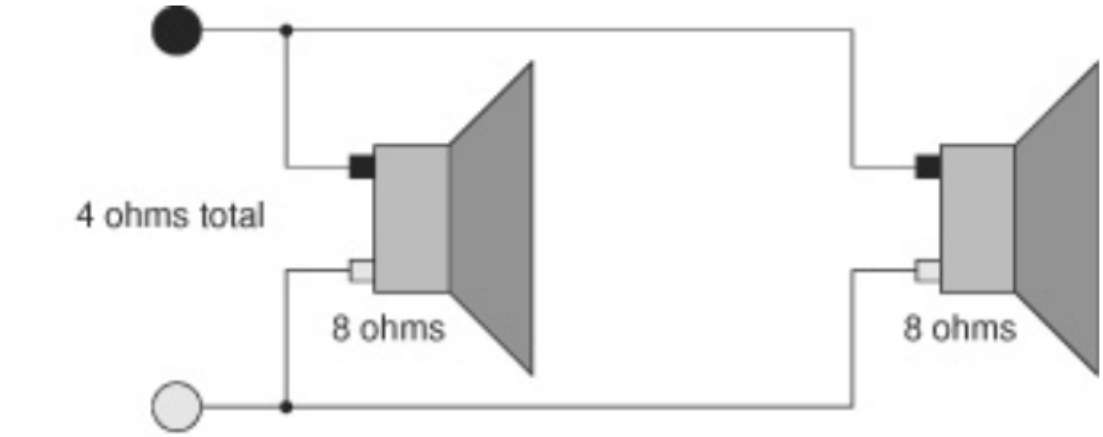
When the electricity passes into the speaker, some of it is resisted. The ohms rating of the speaker is a measure of how much electricity is resisted by the speaker, and an indication of how much energy it takes to drive it - the higher the ohms rating, the more difficult it is to drive.

An amplifier amplifies the electrical signal, causing the speaker to vibrate with greater force; otherwise the impulse would be barely audible (think of the size of a microphone vs. the size of a speaker). How much the amp drives the speaker is measured in watts. When an amp manufacturer gives its wattage rating, it's always at a specific Ohm resistance. "100 watts @ 4 ohms," for example. When the resistance is 4 ohms, this amp will produce 100 watts. If that resistance changes, the amp will produce a different amount of watts.

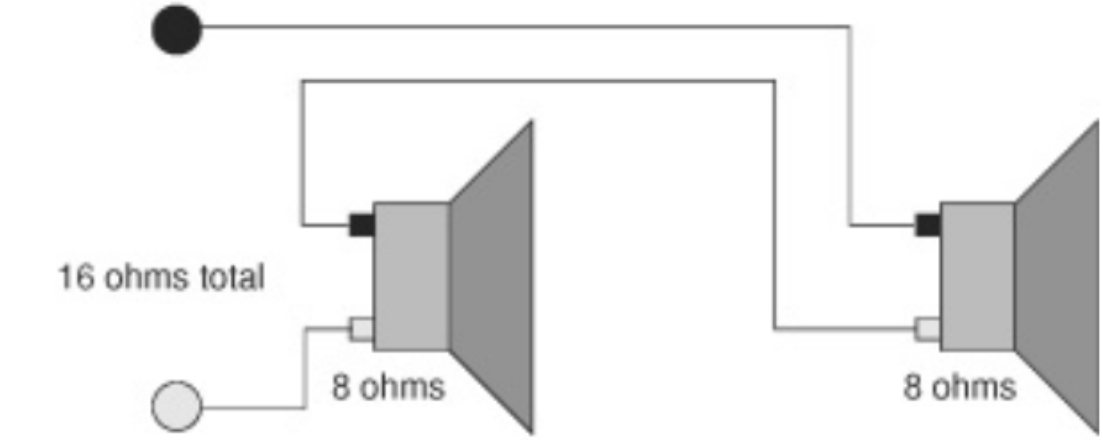
If you're still confused, think of water in a hose. The longer the hose, the less pressure is produced at the end of the hose, even though the same amount of water is flowing through it.
In order to get enough water out of the end of the hose you would have to increase the pressure. In doing so you would have to make sure that the hosepipe could handle the increased pressure.



I = V/R
I: Current
V: Voltage
R: Resistance



Parallel connection



Series connection

Parallel reduces the total resistance increasing the total output. That is, because electricity flows through all of the speakers simultaneously, each speaker added reduces the resistance of the chain. If you visualize all of the negative electrons on one side trying to get to the other, they'll have an easier time because there are so many possible paths for them to go by.

Resistance = (Speaker A x Speaker B) / (Speaker A + Speaker B) Resistance = (8 Ohms x 8 Ohms) / (8 Ohms + 8 Ohms) Resistance = 64 / 16 Resistance = 4 Ohms

Series increases the total resistance, reducing the total acoustical output. That is, because the electricity has to flow through each speaker one at a time, each speaker adds its resistance to the whole. The formula is as follows:

Speaker A + Speaker B = Total Ohms Rating 8 Ohms + 8 Ohms = 16 Ohms

Two 8 Ohms speakers wired in series will have a total rating of 16 ohms.

With this in mind it is important to also use the correct thickness speaker cable capable of carrying the ohms needed.

Mixing Desks

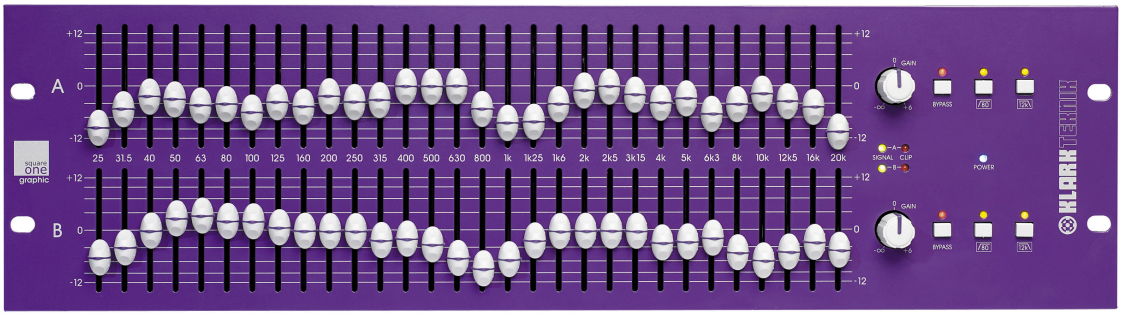
Mixing desks come in two forms: Analogue and Digital.

Analogue Desks



Analogue desks have a button, knob or fader for everything the desk is capable of doing, and it is all laid out in front of you. Analogue desks allow you to adjust basic equalizer (EQ) settings (Bass, Mid and Treble) and sometimes give you the ability to add basic effects (Echo, Reverb etc.). For a more advanced EQ you would have to purchase a graphic EQ to send the sound to before it goes to the amplifier. For more advanced effects, again you would have to purchase an effects unit to send the sound to, and then return it to the desk.

Graphic EQ Unit



Effects Unit



Digital Desks



Digital desks are an amazing tool that combines the basic control an analogue desk gives you, and packs in lots of different built in effects, graphic EQs and many other features.
Digital desks have only one button, knob or fader (lets say BKF for short) for each job it is doing. Analogue desks replicate the same BKF for each channel it has. You can look at an analogue desk and wonder how you would ever learn what each BKF does. But the reality is you only learn say 30 BKFs and the other 200 are repeats for other channels.
Digital desk have only one of each BKF. There is a ‘Select’ button on each channel that once pressed allows whatever you change in its menu or by changing/moving any of the BKFs will only be for that channel that you’ve selected. This results in desks taking up a smaller footprint in physical size, but allows control of even more channels than before.

I could write a Bible’s worth of text about mixing desks, but instead I am going to talk about a few key features to help you understand what happens to the sound when it enters the desk before leaving again.

Desks have inputs and outputs. When you plug in a microphone, instrument, iPhone etc. you want the sound to go into the desk. So you are using an input. When you want the sound to leave the desk to go to speakers, monitors, hearing loop system etc. you are using an output.

Preamp

When sound travels to the desk its level is quite low. Desks have what is called a preamp or pre amplifier. At this point I should say that ‘pre’ means before and ‘post’ means after. So this is a little boost before it gets to the big amplifiers and on to the speakers. The preamp allows you to adjust what is called the gain. The gain increases the volume coming into the desk. Earlier we talked about Phantom Power (+48V). This can be switched on or off at the preamp stage.



On this desk the preamp is on the left. The preamp is used in conjunction with the fader. The LEDs that light up above the fader show the level of sound coming into that channel – not out of it! All sound carries an element of background noise. If the level coming into the desk is too low and you try to then amplify it, you will be amplifying the background noise or hiss with it. If you have a nice level coming into the desk by increasing the gain in the preamp, then you will have a lovely, clean sound being amplified out to the speakers. Too much gain and the level coming into the desk will be lighting up the red LEDs, which means it's peaking. Peaking causes distortion and damage to equipment.

When setting up a desk you should always try to mix with your faders at the 0db level – pictured to the right. Achieving this and getting a good input level is a combination of using the gain and your fader until you are happy. When you are live into a service you shouldn't need to alter the gain. Altering the gain increases or decreases the volume everywhere including monitors on stage.

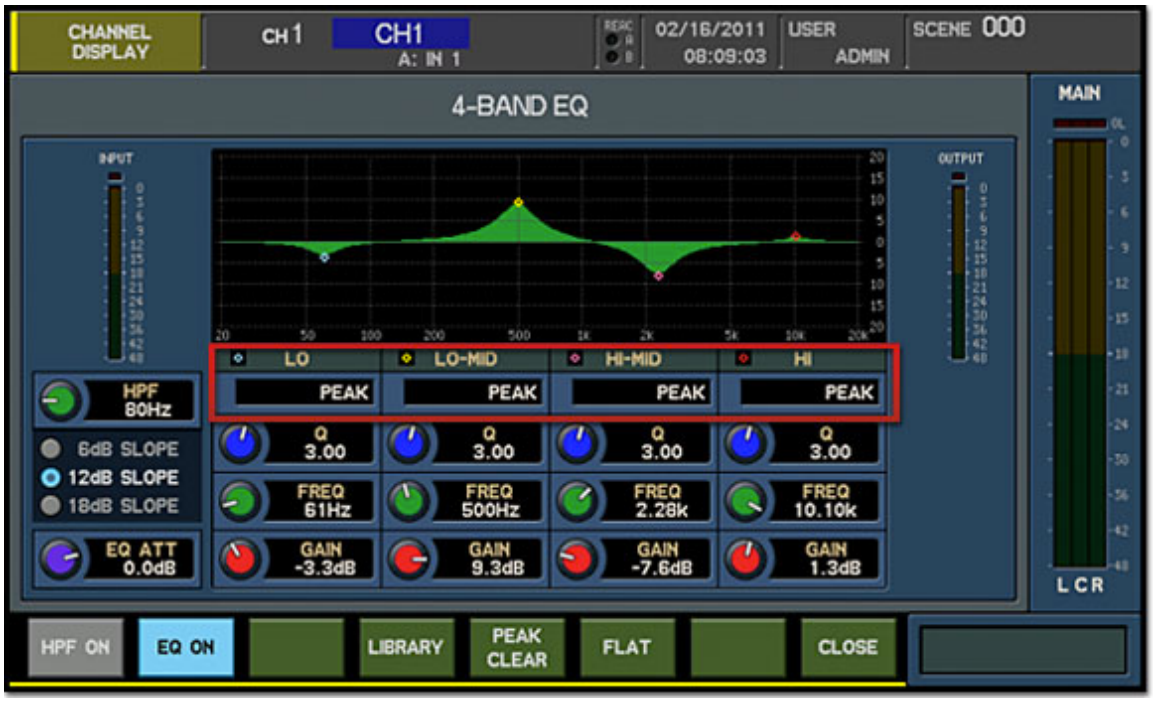


EQ

Once through the preamp, the sound needs to be EQ'd. The EQ can be found here:



Each section of EQ has a gain to increase or decrease, a frequency selector and a Q adjuster. If you press the display button in the EQ section the main display shows you exactly what is going on for that particular channel you are working with.



When you change the gain the graph will increase or decrease in level, meaning you are adding for example more bass or taking away bass.
The frequency allows you to adjust an exact frequency or frequency range.
The Q adjuster allows you to narrow your frequency range down or widen it out. For example if you selected to adjust 125 Hz in your bass range, if the Q was narrow then you would literally only be adjusting around 115 – 135 Hz. If you increased the Q range then you would be adjusting a larger area around the 125Hz you have selected.

Fader

The fader gives you control of the volume going to the front of house speakers (FOH). If you pull the fader down, then no sound comes out of the FOH speakers. Push the fader up and you start to hear the sound. The fader can be used as a control for a number of different features on the desk depending on what section of the desk you are working in. The labels above the fader in the pictured example show that this fader can be used as Channel 1, Channel 25 or the master volume for Aux 1.
There are 3 buttons above the fader. ‘SEL’ is your select button. In order the change EQ, adjust Gain etc. you must select the channel you wish to adjust by pressing this button. Once pressed you cannot un-select the channel. It will remain selected until you select another channel.
‘SOLO’ works if you are using headphones. When you plug in a pair of headphones you hear everything coming into the desk. If you press ‘SOLO’ then you will only hear that channel in your headphones.
‘MUTE’ will mute that particular channel everywhere, monitors included.



Auxiliary

The last section of the desk I am going to talk about in this booklet is our auxiliary outputs – or Aux out for short.



The desk has a number of different outputs. The main output feeds the FOH speakers.
Aux outputs are a way of outputting sound to anywhere else. In our case we use a couple of wedges on stage and in ear monitors. By selecting an Aux e.g. Aux 1 and then pressing ‘SENDS ON FADER’, it turns all the faders on the desk into level controls for what is sent to Aux 1.

So if I had Mic 1 on Channel 1, Mic 2 on Channel 2 and Mic 3 on Channel 3 and I wanted to send the 3 mics down Aux 1 which was feeding a monitor wedge on stage this is what I would do:

Press Aux 1, and it lights up.
Press ‘SENDS ON FADER’, and it flashes continuously.
Now push up channels 1,2 and 3 faders and this now increases the amount of level from the microphones going to the monitor wedge on stage.
Remember to press ‘SENDS ON FADER’ again and it will stop flashing. If you don’t, then any faders you move will only change in the Aux you have selected and not the FOH speakers.

This is as far as I am going to go in this booklet. I hope you found it useful and I haven’t given you a headache!

