FIGURE OF EIGHT

A ZINE FOR WOMXN AND NON-BINARY SOUND ENTHUSIASTS

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00 INTRODUCTION

This is a handbook for anyone interested in sound, looking for an informative resource about recording sound at an introductory level. Omnii is a collective of womxn and non-binary sound enthusiasts, and we're interested in democratising the information and knowledge we've gained from our respective careers as studio engineers, producers, mix engineers, live sound engineers, and musicians.

We wanted to make this handbook for the same reasons that we run our workshops: sound work is a prohibitive and inaccessible field, especially for womxn and non-binary people. At every level of education and employment related to music technology, the number of womxn and non-binary people dwindles the further up the system you go. In our specific experience, it is particularly hard to enter music technology-related education and workspaces, as a direct consequence of our gender identities. The gender imbalance is not only evident, but alarming, and requires immediate action.

We find that looking up audio technology questions online often leads us down a rabbit hole of information, and is often pitched at an inappropriate level for our understanding, either too simple or unnecessarily complicated. Also, the information we find online is almost always written by and for cisgender men. This handbook was written by womxn and non-binary sound workers, for womxn and non-binary people. This handbook is not intended to dumb anything down, over-simplify, or patronise. We want to distribute and democratise the terminology, and evaluate why we use the language we use.

This zine will always be free to view on our website, omniicollective.com. Share it with your friends, start collaborating and creating new ideas.

01 THE BASICS

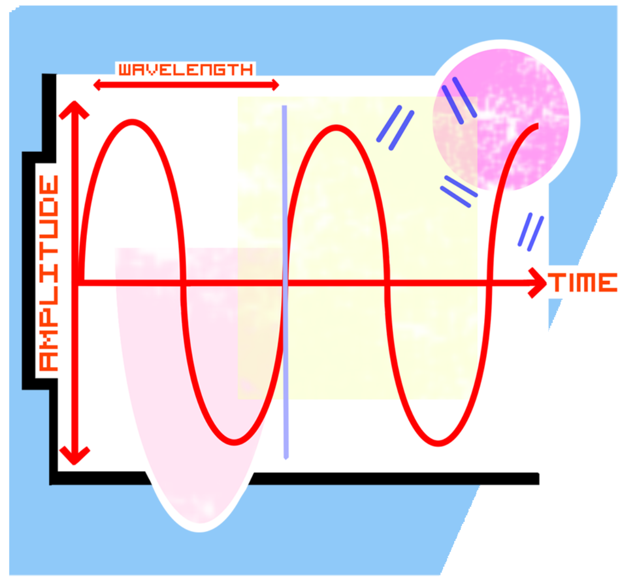
SOUND WAVES

In order to understand how to work with sound, it's helpful to bear in mind what sound actually is: where it comes from, and how it travels from its source to our ears. What we perceive as sound is actually particles colliding with one another, as a consequence of pressure being exerted upon them. Picture a collection of glass bottles, standing closely together in a line. When you knock the bottle on the end, as it topples over with enough force, it knocks the next bottle down, followed by the next one, and so on. When pressure is exerted on an air particle, it knocks in to the next particle, until it reaches your ears and is translated by your ear drum as - you guessed it - sound.

In order to create this pressure in the air, it has to be generated from somewhere. This is typically a vibration: maybe a twanging guitar string or a sub bass speaker cone moving back and forth. As the source vibrates, it exerts pressure onto the air surrounding it, sending the air particles knocking in to one another. The greater the energy coming from the source, the louder the sound will be. The faster the vibration, the higher in pitch the sound will be.

WHEN WE USE THE TERM 'SOUND WAVE', WE ARE REFERRING TO THIS DIAGRAM:

In this diagram, you will notice that the vertical axis is labelled amplitude, and the horizontal axis is labelled time. Along the horizontal axis, you will notice a section marked wavelength. The wavelength determines the pitch, or the frequency.



Amplitude offers a scientific explanation of how we interpret volume, or loudness. In a waveform, amplitude is the vertical measurement between the waves: the greater the amplitude, the louder we perceive the wave. We measure this in Decibels (Db). Decibels, more specifically, are used in sound as a unit of sound pressure level; a high volume exerts more pressure on your ears than a lower volume does. In a live sound space, such as at a gig or in the studio, it is useful to measure the decibel level with a sound level meter, to ensure your listeners are not being subjected to unnecessarily high volumes which could be damaging to their hearing.

The frequency of a sound determines its pitch, or how high or low it sounds when you listen to it. The frequency of a note is found by measuring the distance between each sound wave; the shorter the distance, the higher the note will sound. Frequency is measured in Hertz (Hz). A healthy human ear can perceive frequencies from 20Hz to 20,000Hz.

POWER

A sound source has to have some sort of energy to create a vibration. In acoustic instruments, this energy is generated in a number of different ways, typically as a consequence of the exertions of the human body: as wind instruments require breath, stringed instruments demand bowing and plucking, pianos and drums need hitting. With electric instruments, you'll need... electricity. Your guitar amplifier, studio console, speakers and monitors, all require power to create sound. Incidentally, this is always a useful thing to remember when you get to the rehearsal and your synthesiser - which worked perfectly an hour ago - suddenly won't make any sound. Is it switched on at the wall? Don't worry, we've all done it.

VOLTAGE (V) = [CURRENT (I)] × [RESISTANCE (Ω)]

CURRENT (I) = [VOLTAGE (V)] ÷ [RESISTANCE (Ω)]

RESISTANCE (Ω) = [VOLTAGE (V)] ÷ [CURRENT (I)]

WATTS (W) = [CURRENT (I)] × [VOLTAGE (V)]

CURRENT (I) = [WATTS (W)] ÷ [VOLTAGE (V)]

VOLTAGE (V) = [WATTS (W)] ÷ [CURRENT (I)]

When thinking about electricity, it is useful to understand the terms we use to describe different aspects of electricity, and why we use them. There are three units to think about: voltage (measured in volts), current (measured in amps), and resistance (measured in ohms). These relate to one another in a simple equation: current equals voltage divided by resistance. By extension, we understand that voltage equals current times resistance, and resistance equals voltage divided by current.

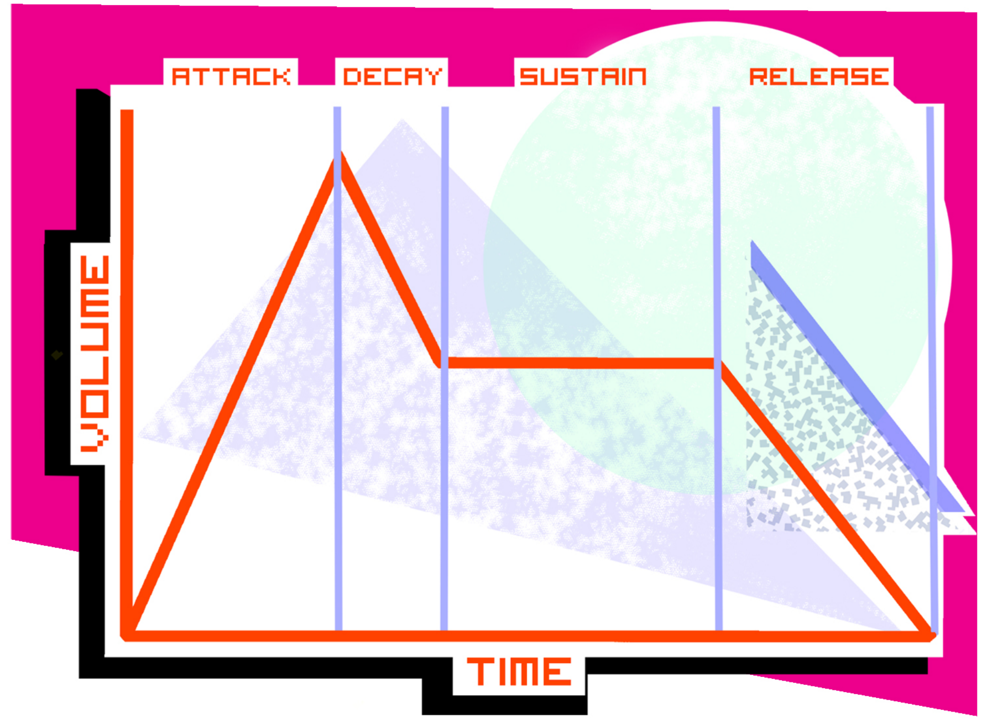
Now, electrical power is measured in Watts. Power is equal to voltage multiplied by current. To distinguish between volts and watts, you should remember: voltage is potential power, watts are actual power. A 50V battery has the potential to deliver power, but it has to be drawn by something. A 150W amplifier can draw up to 150W of power from a source, but it's worth noting that the actual amount of power the amplifier will use will typically be way less than this, unless it is cranked all the way up to full volume, gain, and distortion.

Power is really important to consider in any sound work, whether it's in a music venue or a recording studio. It's not just important for the quality of sound, but it's also essential in keeping you (and your equipment) safe.

Power from a typical 3-pin wall socket in your home will draw a current of 13 amps. This is important to keep in mind, to make sure you do not overload a socket. To check how much power you can draw from a single socket, you can use the P=iv equation illustrated above. The mains voltage in the UK is 230 Volts, so if we multiply that by 13 amps we have 2990 Watts of power we can draw from the wall socket. You can usually find the wattage of your various appliances on a sticker near the power socket, so simply add those together and check that the total does not exceed 2990 Watts.

ADSR

ADSR is an acronym for Attack, Decay, Sustain, Release. The ADSR of a tone provides the sound with sonic character. The attack, decay, sustain, and release of a tone measure the time it takes for the sound to reach different points in its formation, from the beginning to the end of its journey.



Attack reflects the time from the initiation of a note to reaching its full amplitude, or loudness. The attack period begins when the note begins and ends when it reaches its loudest volume. A lengthened attack can sound like a note swelling in size.

Decay is the measurement of the time between the full loudness of a note and its descent to the sustained level of volume. The decay period begins when the note is at its loudest, and ends when it reaches the level it will remain at for the sustain period.

Sustain indicates the time between the sound reaching its sustained level, to the key being released. Sustain begins when the volume of the note stabilises, and ends when the key is released and the volume begins to fall.

Release is the time between a note being released, and the sound disappearing. Release begins when the note is let go, and ends when the note is inaudible and no more signal is being passed.

02 TRANSPORTING SOUND

CABLES

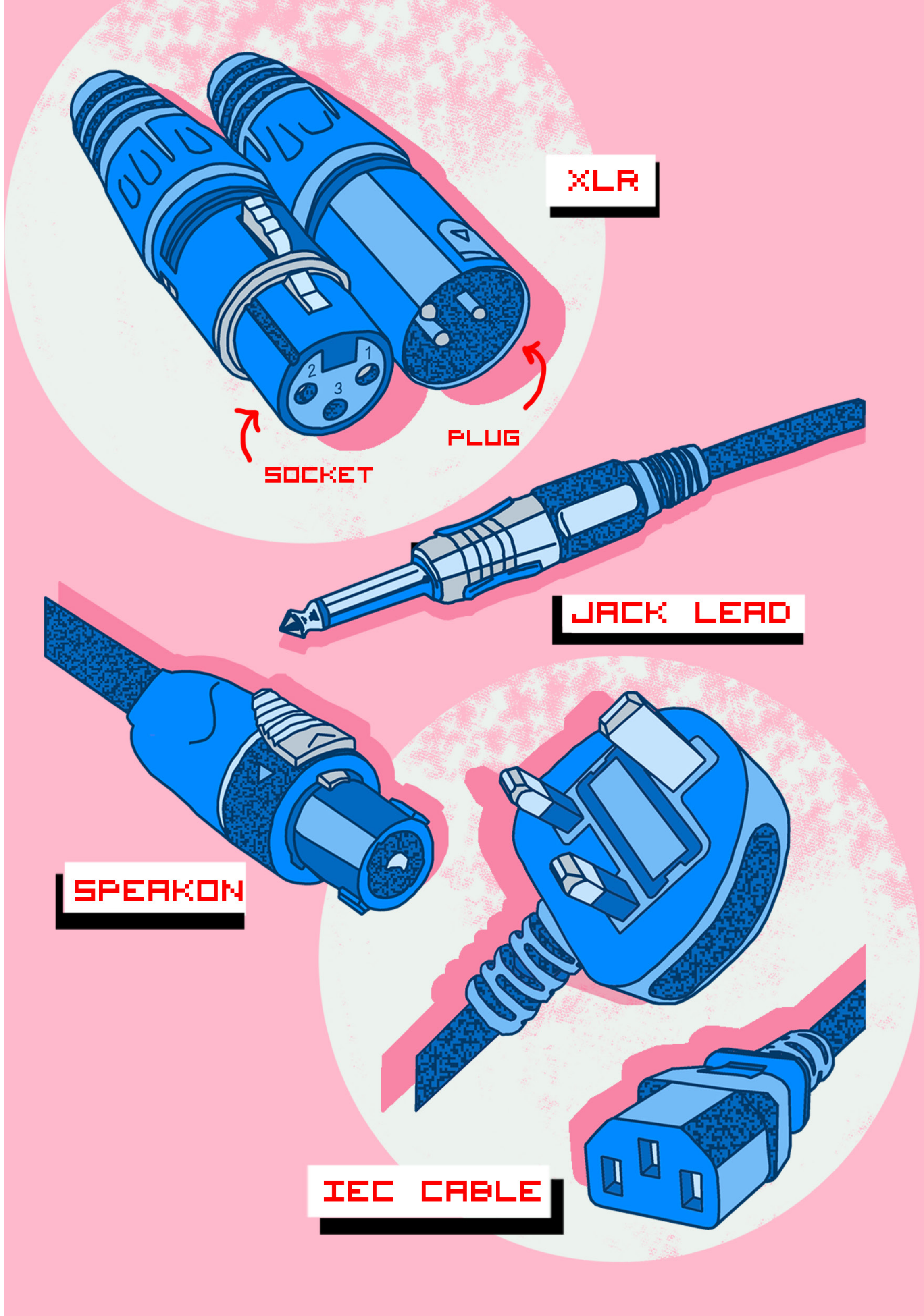
Mostly, you will encounter two types of cables: XLRs, and jack leads. XLRs are used for most microphones, and jack leads are used for instruments such as synthesisers, keyboards, guitars and bass guitars.

That may well be as much as you'll ever need to know about cables. However, as sound workers and musicians, we use a much wider variety of cables than just the two above, and knowing the nuances of what goes where can be really useful, not only for ensuring the quality of your sound, but even for making sure you're not damaging any of your equipment. It's easy to tie yourself in knots, literally and figuratively, so here's a quick breakdown of what to think about when you're plugging something in.

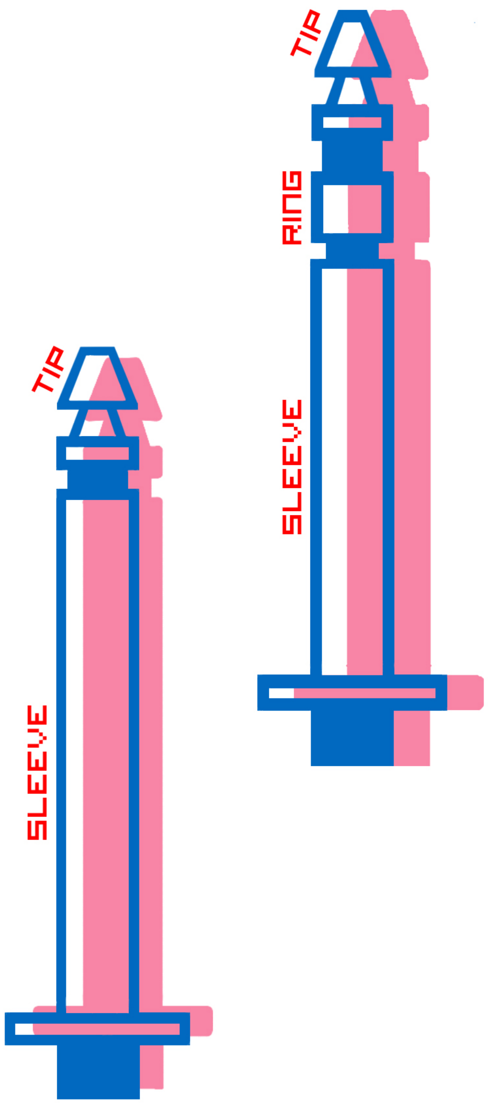
First of all, we need to think about power: does the instrument or speaker you're plugging in require power to make sound? Is that power provided by a specific power only cable, or a combination cable that transports both power and sound? An example of a power only cable is an IEC: An IEC cable, also known as a kettle lead, is a power cable that connects a device to the mains electricity supply. You'll use an IEC cable with many amplifiers, mixing desks, keyboards and synthesisers.

Some devices are described as active or passive, depending on their relationship to power. A piece of equipment would be described as 'active' if it requires direct connection to a power source to produce sound, such as via an IEC cable, or kettle lead. Once it receives this power, the device is able to send or receive audio without any additional amplification. A 'passive' speaker or DI requires an additional stage of amplification before it can send or receive signal. For example, a lot of hifi speakers require signal to be sent via an amplifier to produce sound from a signal sent by a turntable.

Some cables send both power and audio signal. A common type of cable that does exactly this is a speakon cable. You'll usually find speakon cables plugged into a PA and monitors.



Going back briefly to jack leads and XLRs, it's necessary to distinguish between balanced and unbalanced cables. Because audio is an AC (Alternating Current) signal, every audio cable has two conductors inside: one positive and one negative. A balanced cable has an additional grounding wire inside. An unbalanced cable uses the negative conductor as a ground simultaneously. All XLRs are balanced cables, which explains why they have three pins in the plug connector. Jack cables are typically unbalanced, but some instrument cables will be balanced - you can determine the difference by checking for a ring, tip and sleeve.



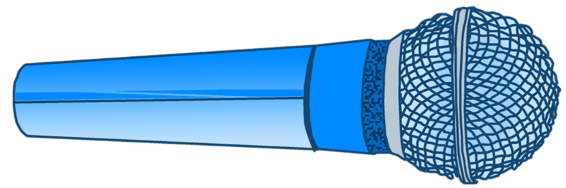
Quite reductively, the terms 'male' and 'female' are crudely used to describe the function of each end of a cable. 'Male' cables usually send signal, and 'female' cables receive it, for example: if you want to plug a microphone into your audio interface, you plug the female end into the microphone to receive the signal, and the male end into the interface to send that signal to your DAW. Pointlessly gendered items such as these are exactly why we wanted to create this handbook, as they bring us to an era where gender could be reduced to a sum of two different parts. This is not how we know gender to be, and equipment should not be referred to as such, as it demeans everyone no matter how you identify. Instead of 'male' and 'female', you can use plug and socket to refer to XLR cable ends.

It sounds obvious, but before plugging in you need to consider what is your input and output. For instance, if you're rigging up your own pedal board, you need to make sure you get your ins and outs the right way around, or else you'll get no sound.

Also, this is important for performers: when you show up for sound check, a professional venue will typically provide all XLRs, but you'll need to bring your own jack leads, as most venues won't provide them.

MICROPHONES

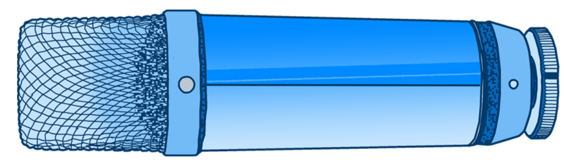
There are several different types of microphone you will encounter, both in the studio and in live sound. Different microphone types are suitable for different uses, however it's worth noting that these are just suggested uses, not hard and fast rules. If you like how a particular microphone sounds with a particular instrument, then go for it!



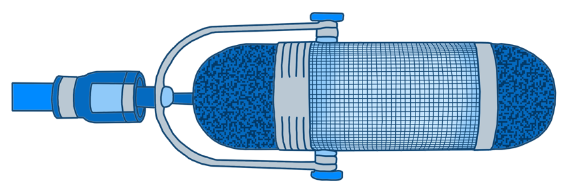
DYNAMIC

A dynamic microphone contains a coil within a magnetic field. When sound waves enter the coil, it causes the coil to vibrate within the magnetic field, generating an electric current. Because the dynamic microphone creates its own current using magnetism, it does not require phantom power like a condenser microphone.

CONDENSER



A condenser microphone is a type of microphone in which a small voltage is passed between two plates, one acting as a very thin diaphragm. When sound waves hit the diaphragm, the distance between the two plates decreases. This is measured by a capacitor. Because of the voltage involved in the process, condenser mics require phantom power to be used. Condenser mics are extremely sensitive, and excellent for studio recording; however, because of the sensitivity they are susceptible to feedback, which makes them more challenging to use in a live sound environment.

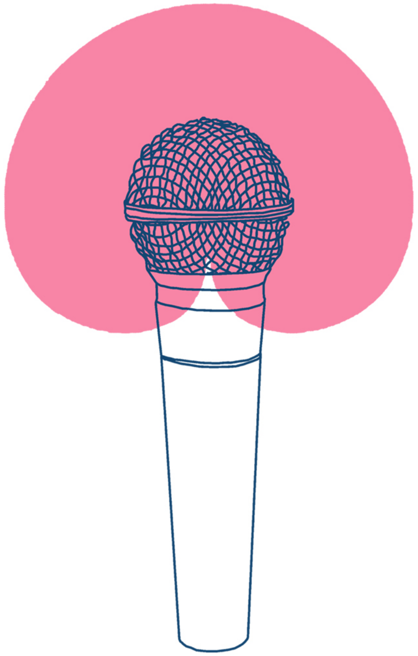


RIBBON

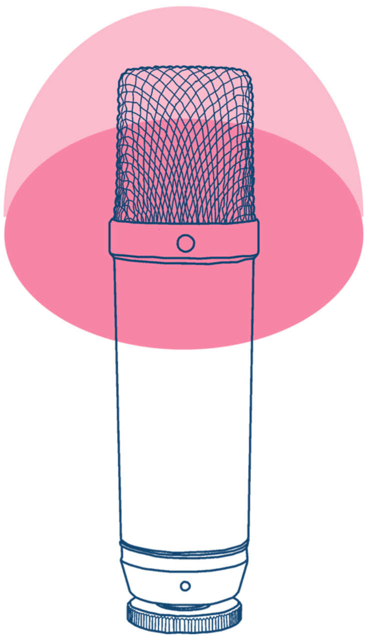
A ribbon microphone is similar to the dynamic microphone, as in it creates its own magnetic field and does not necessitate the use of phantom power. Inside a ribbon mic, there is a thin strip of metal within a magnetic field. The ribbon combines the diaphragm element of the condenser with the transducer of the dynamic microphone. Ribbon mics have a figure-of-eight polar pattern. This is important: you absolutely must not use phantom power with a ribbon microphone, as it will damage it. Ribbon microphones can also be damaged by air gusts - so when using ribbon mics on guitar amps, kick drums, horns or vocals, the microphone should be placed much further away from the soundsource than you would normally place a traditional dynamic or condenser microphone. The ribbon inside the microphone is extremely thin, which makes it excellent for picking up subtle nuances in sound, but unfortunately makes them fragile. Handle with care!

The polar pattern of a microphone is a diagram which demonstrates the direction from which the microphone receives sound from. Many condenser microphones have switchable polar patterns, making them useful for many different applications. There are three common polar patterns: cardioid, omnidirectional, and figure-of-eight. Cardioid is a heart-shape, which records only what is in front of the microphone. Figure of eight records both in front and behind the microphone. Omnidirectional records all around the microphone, and, incidentally, inspired the name of Omnii Collective.

CARDIOID



OMNIDIRECTIONAL



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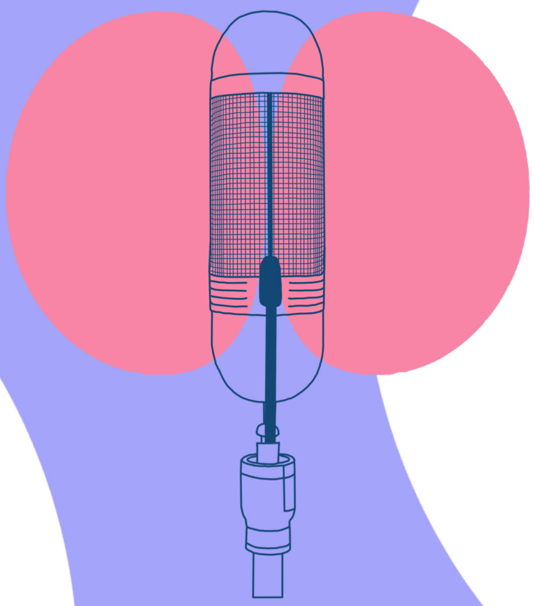


03 THE MIXING DESK

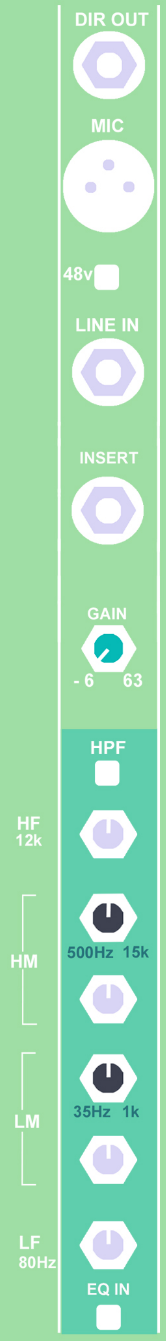
DIGITAL AND ANALOGUE

The mixing desk is used in recording studios and in live sound to receive and adapt audio signals, whether it's getting the perfect level for your vocal recording, or mixing a live gig. In the studio and in live sound, you will encounter a wide array of mixing desks used for lots of different applications. There are two main types of desks you will need to be aware of: digital desks, and analogue desks. Analogue desks transport information via a series of electrical pulses, whereas digital desks use binary code.

When you see a recording studio in a film or on TV, you'll most likely see a massive analogue desk with a thousand knobs and buttons. What's important to remember when working with an analogue desk, is it actually contains several copies of the same channel strip, so the thousands of knobs aren't actually as intimidating as they seem.

Most analogue desks work from the same principles, and are usually arranged with a particular look and feel. On the next two pages you'll see a detailed annotated diagram of the channel strip, which is taken directly from the Allen and Heath ZED desk. The concepts and functions of this channel strip are pretty universal to any analogue desk.

Digital desks have a vastly different layout, as owing to the nature of the technology they use, the copying of the same strip of knobs and buttons several times over simply isn't necessary. You'll probably still find the faders towards the bottom, and a few options for assigning different input, output, and monitoring channels to each fader. The knobs and buttons for EQ, rather than being copied several times over, are usually grouped together at the top, and you'll have to select each channel individually before adjusting.



Direct Outputs - These send the signal of the channel straght out to a source of your choice and can usually be set pre or post fade. We're using these to send the signal from the mixer into the Audio Interface.

Mic Input - Balanced XLR input

48v (phantom power) - This button sends voltage through the cable and is used to power condenser microphones and some DI boxes. 48v can break some microphones such as ribbon mics so check your set up before sending it.

Line In - Input for 1/4" jack cables. On this desk both balanced and unbalanced can be used.

Insert - Used for 'inserting' hardware effects such as compressor, limiter, noise gate, into the channel. When a connection is made via the insert point, the signal is routed via the external device and then back into the channel to continue down the channel strip. When nothing is plugged into it, the signal is routed as normal. Inserts usually come before the channel EQ section, though on some mixers it can be switched to pre or post EQ.

Gain - Raise the gain to see signal from the mic coming into the channel. The desk we're using starts at -6 and goes up to 63.

EQ section. Use this to boost or cut frequencies.

Cut or Boost (-15 to +15 dB) HIGH FREQUENCY

Frequency Select (500hz - 15K)

HIGH MID FREQUENCY

Cut or Boost (-15 to +15 dB)

Frequency Select (35Hz to 1k)

LOW MID FREQUENCY

Cut or Boost (-15 to +15 dB)

LOW FREQUENCY

Cut or Boost (-15 to +15 dB)

EQ In button - Activates the EQ

The Aux section (or Sends) - Auxiliaries (known as auxes) are used to send signals to external effects, headphones for hearback, or monitors in a live gig situation. They are multi-functional and you can be creative with how you use them.

Auxes 1 - 2 are pre fader, this means the level of them is independent of the volume fader. Therefore you can raise and lower the volume fader without affecting the level you are sending. This is very useful, especially when you are sending to musicians' headphones and don't want your changes to affect what they hear.

The pre/post button allows you to choose whether you want to send the signal pre or post fader to Aux 3 and Aux 4

Auxes 5 and 6 are post fader, this means the fader will effect the level you are sending out of the aux. Post fader auxes are most commonly used for reverbs.

Pan - sends the signal to the left or right of the stereo field.

Mute - mutes the channel.

PFL (Pre Fade Listen) - check the audio signal before raising the fader or unmuting the channel.

The signal and peak LEDs let you know when there is signal coming in and when the channel is clipping.

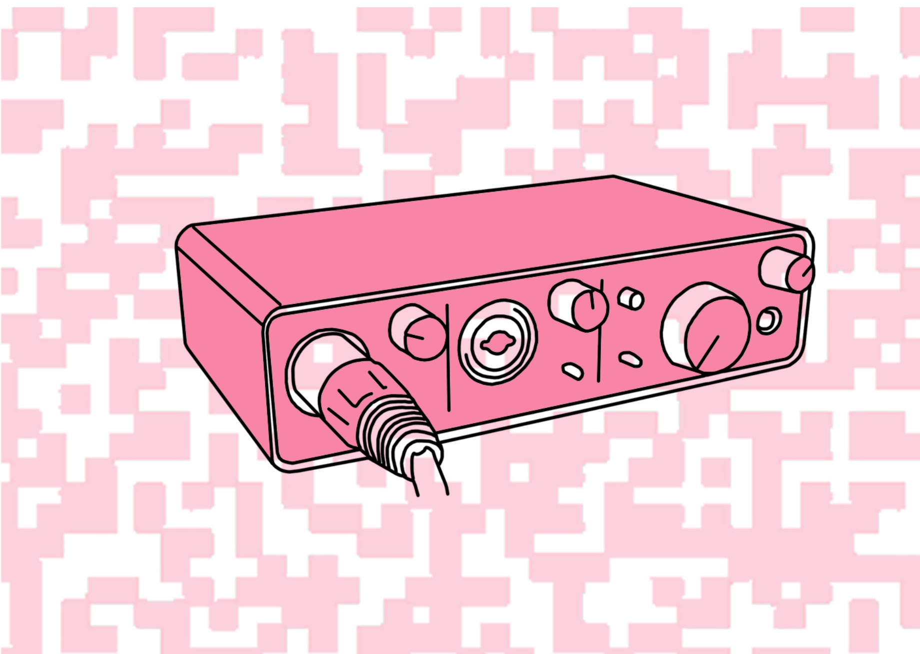
The routing switches connect the signal to the mix buses (left & right, mono, and the 4 group busses).

Volume fader - raise the volume of the channel to send signal to the left & right, mono and group buses and auxes 5 & 6, also auxes 3 & 4 if switched to post-fade. The unity gain position is marked by "0".

* This diagram is modelled on an Allen and Heath ZED mixer, but the same concepts apply to all mixing desks.

INTERFACES

If you're recording into a computer, you'll need an audio interface to convert the signal from your microphones and instruments into a signal your computer can read. Audio interfaces convert the analogue signal sent via an XLR or jack cable, into binary code. There are a wide array of interfaces out there, with different numbers of channels. In a studio with an analogue desk, you'll probably see the channel and group outputs of the desk connected to an audio interface, which sends the desk signal into the computer for recording and editing. We'll cover audio interfaces again in chapter 7, when we look at bedroom setups for recording.



04 PROCESSING

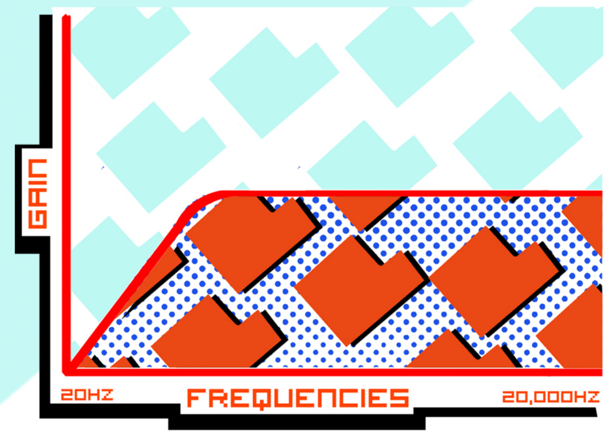
EQ

EQ is a dynamic process allowing you to sculpt your sound. You can control how loud or quiet certain frequencies, or groups of frequencies, will sound. This is a handy feature to use on individual channels within a track, but also over groups of sounds, as EQ can make a lot of difference in how we perceive sounds.

For example, boosting the high frequencies of a sound source can make it sound more bright and shimmery. If a track is sounding muddy, you can try cutting some of the mid frequencies. If something needs more or less bass, you can cut or boost that too. EQ can also allow you to sweep through and find any 'problem' frequencies that might be ringing out too much, or clouding the sound of the instrument in some way.

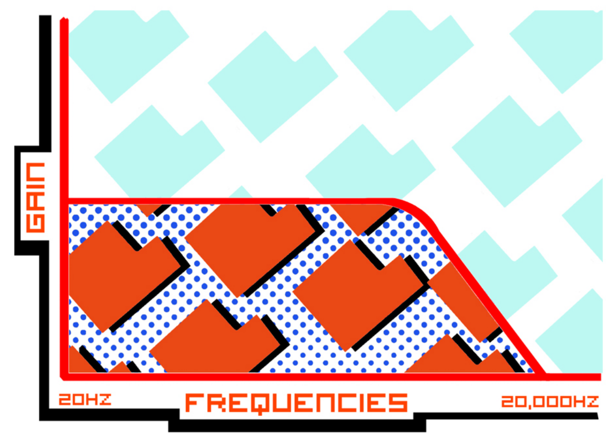
HIGH PASS/ LOW CUT FILTER

A high pass filter is a feature of EQ. It allows high frequencies to pass through, whilst cutting bass frequencies.



LOW PASS/ HIGH CUT FILTER

A low pass filter allows low frequencies to pass through, whilst cutting high frequencies.



COMPRESSION

Compression is the process of reducing the dynamic range between the loudest and quietest parts of an audio signal. This is done by boosting the quieter signals and attenuating the louder signals. Picture the waveform of your audio in your mind: imagine taking the quietest parts and stretching them out, and taking the loudest parts and squishing them in. Every compressor, whether analogue or digital, is different, but all contain variations of these functions:

THRESHOLD

A compressor's threshold sets the level of compression. With the threshold at maximum, very little is is compressed; when set it to minimum, most of the signal will be above the threshold, so the level of all but the very softest bits will be reduced (or compressed).

RATIO

Affects the amount of compression. It sets how drastically the compressor reins in signals that overshoot the threshold level. 4:1 is the lowest ratio, 20:1 is the highest. A 4:1 ratio means if there is an increase of 4db in the input signal, the compressor will only output 1db.

ATTACK/ RELEASE

Attack is the time it takes for the compressor to respond to incoming signal. Release is the time it takes for the compressor to return to the original signal level. Fast attack time means gain reduction kicks in almost immediately and catches transient signals of very brief duration, reducing their level and 'softening their sound'. Slower attack times allow transients to pass through unscathed before limiting or compression kicks in. So if you want to keep the initial punch of a sound, a slow attack is the one to choose. If you want to squish the initial impact of a sound choosing a fast attack will do this.

MAKE UP GAIN

After applying these processes, your overall level will be quieter, as you have taken down the peaks in the audio. Therefore, you can use make up gain to bring the signal back up to the required level.

KNEE

Sets the strength of compression at levels close to the threshold. Lower values result in more severe or immediate compression (hard knee). Higher values result in gentler compression (soft knee).

GATES

Gates are another type of dynamic processing which are used to take away unwanted sound from a signal and used heavily in live engineering, studio engineering and mixing. A gate works just like a compressor, except instead of reducing the peaks of a signal, a gate only allows signal to be heard once it exceeds a certain amplitude. You can set this level via the threshold, just like you do with a compressor. For example, a mic on a snare drum will often pick some signal from the other drums on the kit, especially those closest, the kick drum and hi-hat. If you want to reduce the level of these, you can set the gate threshold so only the loudest peaks (ie. the hits on the snare itself) are allowed through.

05 EFFECTS

REVERB

Reverb is an effect used in audio production to alter the space in which a sound is heard. When a sound is played into a space, it reflects off the walls, meaning that the sound can be heard for a length of time after it has finished being produced. Think of the difference between singing in a carpeted bedroom, and a church hall; in a larger space with stone walls that can't absorb any sound, your singing will sound more full and luscious, as it will reflect and resonate within the space; the reverb is extremely different. Digital reverbs, such as plug-ins on Logic Pro, can digitally recreate a space in which to send signal, to create the illusion of it being played in a space.

DELAY

Delay is an audio effect in which a sound is played back in a staggered manner, several times over, following the sounds' initiation. The effect can either be played back separately from the original audio source, or played back within the original source, creating feedback, in which the source hears and repeats the already repeated material.

Delay and reverb are really useful audio effects, as they add depth to your sound.

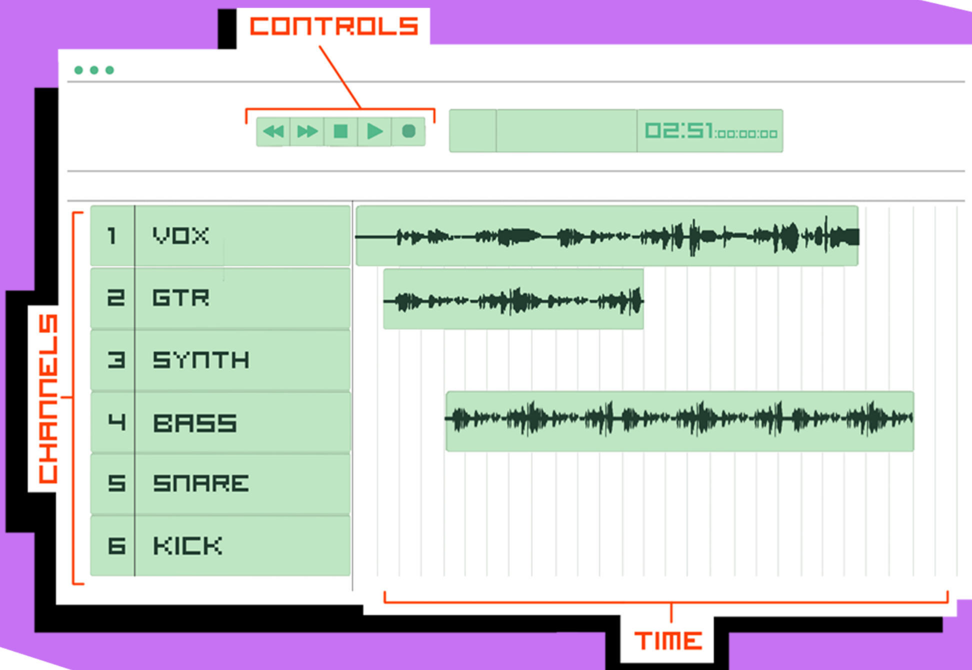
06 DAWs

LOGIC PRO, ABLETON LIVE, PRO TOOLS

A Digital Audio Workstation is a computer programme designed for recording and editing digital audio. It's what you'll use when you want to record and produce some music.

There are a number of different DAWs you will encounter, with varying qualities and price points. Using free softwares such as Audacity and Garageband is a great way to learn about the basic functions of these programmes, as they enable you to record and use basic editing functions to craft and sculpt your sounds. There are three DAWs that will be discussed below: Logic Pro, Ableton Live, and Pro Tools. There are plenty of other DAWs, such as FL Studio, Cubase and Reason, which operate in a similar way and are great to use for recording and live set ups.

Most DAWs use a horizontal timeline interface, so the audio recording starts on the left hand side and moves to the right, much like reading sheet music. Each different channel is stacked on top of one another in a vertical list.



PRO TOOLS is often regarded as the industry standard for studio recording. Whilst the software on its own doesn't allow for much experimentation with software instruments or digital effects, the user interface essentially acts as a blank slate for you to lay out your ideas and cut and paste samples and clips wherever you would like.

LOGIC PRO is essentially the big sister of Garageband. It's made by Apple, and only available to use with Mac software. Logic is great if you're looking to do a bit of a combination of using MIDI instruments and recording live audio, as it has a great selection of included software instruments and audio effects, as well as a number of plug-ins produced by external companies that you can use within Logic to sculpt your sound. It's also super easy to use when recording, editing and layering audio.

ABLETON LIVE is slightly different to other DAWs, as it is engineered for DAW-assisted live performances. It has two different modes of interface: the horizontal timeline, as with Logic Pro and Pro Tools, and a vertical interface in which the channels and samples are stacked vertically. Ableton is great to use with MIDI-enabled hardware for live performances and recording, such as sample pads and keyboards.

MIDI

MIDI is an acronym for Musical Instrument Digital Interface, the language through which a controller, such as a MIDI keyboard, can communicate digitally with other devices. For example, when connecting a MIDI controller to a software instrument in Logic Pro or Ableton, MIDI is what enables the computer to understand the signal from your device, and allow your device to control the software.

07 GET STARTED!

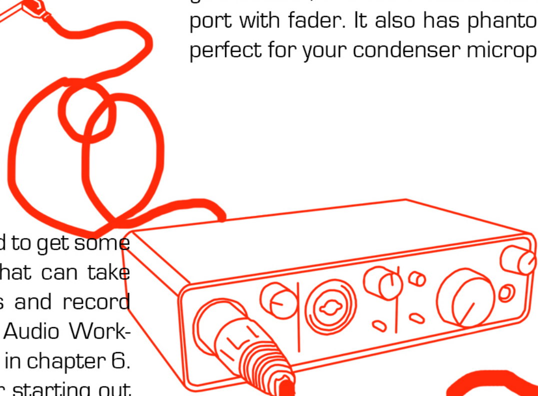
BEDROOM RECORDING

So! Newly equipped with all your knowledge about sound, you've got your instruments in hand and a head full of ideas, and you're ready to start recording. In this chapter, we're going to look at all the things you'll need to start recording your own music.

First of all, you're going to need a microphone to record yourself with, if you're planning on recording any live audio. It's worth thinking about what exactly you want to record, and in what style. If you want crystal clear vocals and acoustic instruments, a condenser microphone may be the best place to start. If crystal clear quality isn't your priority, or if you're mostly recording guitar amplifiers or drums, a versatile dynamic microphone may be a good starting point.

For condenser microphones, the RODE NT1-A bundle is a great starting point. The NT1-A is a condenser microphone, with a cardioid polar pattern. It's great for recording vocals, especially in higher registers, as it has a pleasantly bright response. The bundle includes a pop shield, which is used for keeping any harsh pops or hisses out from your vocals. For dynamic microphones, the Shure SM58 is the industry standard for vocals. It's relatively cheap as microphones go, it's almost indestructible, and it's extremely reliable in delivering a great sound. Also, if you unscrew the cap, you have an SM57, which is great for recording instruments such as drums or guitar amplifiers.

Once you've got your microphone, you'll need an audio interface. We covered exactly what these do in chapter 3, but in case you want a reminder, audio interfaces convert the audio signal from your microphone or instruments, and turn it into digital code (binary code, to be precise) which your computer can read. A great little interface to start with is the Focusrite Scarlett 2i2. It has two XLR/Instrument line inputs with gain control, a monitor fader and a headphone port with fader. It also has phantom power, so perfect for your condenser microphone.



From your interface, you'll need to get some software on your computer that can take your binary coded waveforms and record and play them back: a Digital Audio Workstation, covered in more depth in chapter 6. Some great free softwares for starting out with recording are Audacity, and Garageband.

LIVE SETUPS

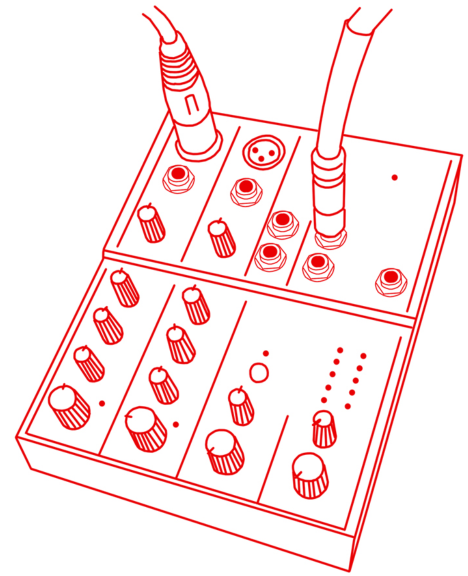
Once you've got a couple of songs recorded, you might be thinking about trying out some of your material live at a gig - or, indeed, vice versa! Crafting your own live setup is an extremely personal process, and it really depends on what type of music exactly that you want to make. Look around at the instruments you have, and the things you want to make music with: how (if at all) do you want these sounds to be processed? Do you want to record/playback anything live? There are a couple of different tools that can be great to have in your live music arsenal.

Loop pedals are a great way to take making music on your own to the next level. You can record and playback instruments or vocals live, and layer up sounds to create a thick texture. There are lots of different loopers that can be used for different things; some have a variety of selective looping options, others may contain effects that you can use on your vocals and instruments.

Sample pads and MIDI keyboards are a great way to take your in-the-box production into a live setting, as they allow for an element of physical control to your performing. There are a lot of models of MIDI controllers by Novation that are made specifically for Ableton Live, which allow you to add dynamic gestures to your live performances.

If you're thinking about using your laptop to send out signal in a live performance, it's worth investing in an audio interface. An interface will create another layer of gain staging between your laptop and the mixing desk, and allow you to more easily control the level you are sending out.

It's worth saying that we're not sponsored by any of these brands (unfortunately!), and there are plenty of other options out there for all of these equipment suggestions above. We recommend doing your research, and thinking long and hard about what exactly you want your equipment to be able to do, before investing. Also, there are a number of ways to get good quality equipment at more affordable prices. Don't be afraid of buying second hand gear; Gumtree and Ebay are always worth checking out, and there are a number of buy / sell / swap sites and groups on social media. Also: don't let a lack of equipment limit you! There are lots of different ways of recording just with your phone, or your laptop, and no other equipment. There are also lots of analogue ways of going about this process: try using an old tape player/recorder, for example.



... And that's it! We've come to the end of our handbook. Hopefully you've learned something you didn't know before, or understood a process you've used a thousand times in a different way. If you have any questions, or anything you would like to explore but don't know where to start, you can join our Omnii General Forum and connect with like minded womxn and non-binary people, or maybe come to one of our workshops! We wish you luck with your sonic endeavours. Enjoy!

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