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MODULE 3

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INTRODUCTION TO TECHNOLOGY (SOUND)

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UNITS

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1. AUDIO SIGNAL FLOW

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The Recording Studio Audio Signal Flow

What happens to sound once it enters the microphone?

It travels through a complicated chain of equipment and eventually re-emerges out your studio monitors/headphones.

That's the short version….

But what's the long version of that story? Where EXACTLY does the sound go? And WHY?

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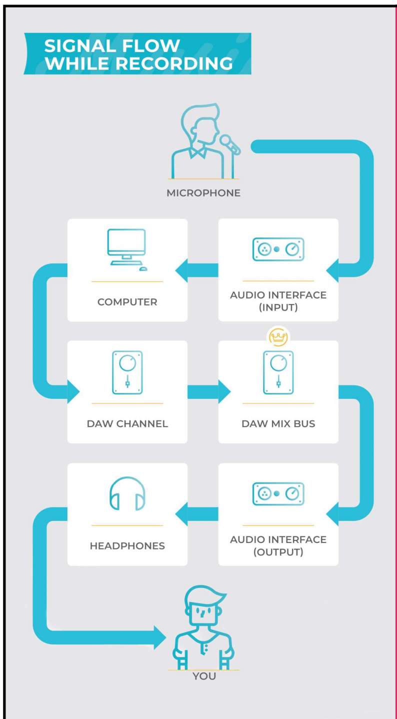

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Are ALL studios like this?

In the most complex studio setups, this is the path an audio signal must travel to reach your ears.

In simpler studios, you might imagine that the signal flow would be simpler as well. The truth is…it's not. It's the same.

A step by step walk through the diagram

1. Microphone-> Audio Interface Input (Mic Preamp)

The microphone picks up the sound, and a *mic level* signal is sent to the microphone preamp. Since mic level signals are inherently weak, the preamp is needed to amplify it to a higher level. The amplified signal is known as *line level*.

- Within the audio Interface Mic Preamp->A/D Converter**:** The mic preamp sends the processed *analogue* signal to the A/D converter to be translated into a *digital* audio signal.

2. A/D Converter-> Audio Interface-> Computer

The A/D converter sends the digital signal to the audio interface, (USB, Thunderbolt, Firewire are some of the protocols used) where it is sent into the computer to be processed by the *Digital Audio Workstation (DAW – Software, i.e. Reaper)*.

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3. Computer-> DAW (All within the computer)

Within the DAW, the signal is processed by any number of inserted plugins (EQ, Compression, Reverb, etc), and mixed with any other tracks in the session.

4. DAW input-> DAW Channel

The channel is where you will process the signal by adding plugins to enhance the tone and clarity of your audio.

5. DAW Mix Bus-> Computer

The signals from multiple channels are summed together to create a stereo signal. Further processing may occur at this point to perform some rudimentary mastering.

6. Computer->Audio Interface->D/A Converter

After all DAW processing is complete, the signal is sent out to the audio interface and passed to the D/A converter, where it is converted back into an analogue signal.

7. D/A Converter->Headphone Amp/Monitor Management The D/A converter sends the new analogue signal to one of two places: either the headphone amp, or monitor management system. This is the final step in the process before converting the signal back into sound.

8a. Headphone Amp->Headphones If and when the analogue signal reaches the headphone amp, it is then sent to the headphones, where it is heard by the performer.

8b. Monitor Management System->Studio Monitors When the analogue signal reaches the monitor management system, it is then sent to the studio monitors, where it is heard by the sound engineer.

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Two facts of supreme importance to keep in mind:

1. The mic preamp is before the A/D converter. So, set your recording level via the mic preamp. If you cause distortion by overloading the A/D converter, turning down the track fader (within the DAW) won't fix the distortion problem.
2. The fader is after the hard drive (in the computer where the audio file is recorded). As a result, the track fader does not affect your recording level.

Here is further information on how to properly set you gain stages in your DAW.

- https://iconcollective.edu/gain-staging/

Conclusion

Of course, there are a many signal flow variations, especially if you start using analog outboard gear or plug-ins in DSP-powered mixers. The good news is that if you feel solid and confident about signal flow, you can run sessions more effectively, and successfully create more complex configurations to suit your creative needs. Be comfortable with the basics first, then get fancy later on!

2. MICROPHONE TYPES & APPLICATION

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How Do Microphones Work?

A Basic Moving Coil Microphone

Microphones are a type of *transducer* - a device which converts energy from one form to another. Microphones convert acoustical energy (sound waves) into electrical energy (the audio signal).

Different types of microphone have different ways of converting energy but they all share one thing in common: The *diaphragm*. This is a thin piece of material (such as paper, plastic or aluminium) which vibrates when it is struck by sound waves. In a typical hand-held mic like the one below, the diaphragm is located in the head of the microphone.

When the diaphragm vibrates, it causes other components in the microphone to vibrate (coils of wire attached to the diaphragm). The vibrations (movement) of these coils in a magnetic field create an electrical current which becomes the audio signal.

Note: At the other end of the audio chain, the loudspeaker is also a transducer - it converts the electrical energy back into acoustical energy.

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Dynamic & Condenser Microphones

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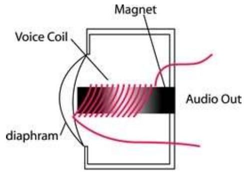
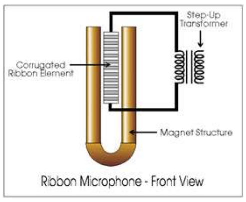
Your **standard handheld dynamic microphone** features a bar magnet wrapped in a coil. Vibration from sound waves causes fluctuations in the magnetic field.

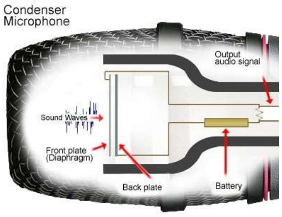


Dynamic Microphone



Ribbon microphones are also dynamic, but feature a rippled strip of metal situated between the ends of a U shaped magnet. Ribbons are more sensitive but not as rugged. There are some handheld dynamics rugged enough to pound nails with (I'm exaggerating, don't actually do this) but ribbons can be a bit delicate.

Condenser microphone diaphragms feature a fixed plate and a freely vibrating one. The diaphragm of a condensor doesn't generate an electrical charge all on its own, so it needs 48 volt phantom power to produce a signal (except for tube microphones which have their own power supply). Some condensers feature built-in batteries and do not require phantom power.



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Introduction to Microphone Polar Patterns

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When discussing microphones there are a number of pickup patterns, and you see the same three words appear over and over. Cardioid, Omnidirectional, and Figure-8. But what do they actually mean? And why are they so important when choosing the right mic?

A microphone's **polar pattern** is the 3-dimensional space surrounding the capsule where it is most sensitive to sound.

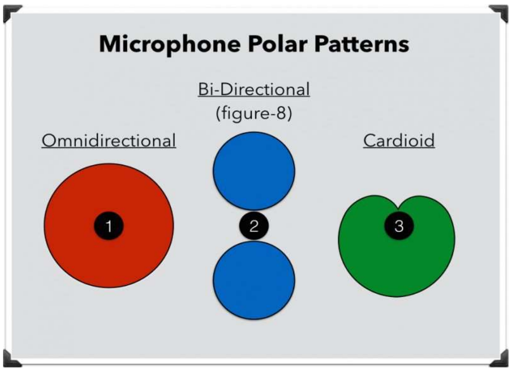
The 3 basic patterns are:

- omnidirectional
- figure-8
- Cardioid

Here's a diagram showing how they look:

As you can see,

- Mic 1 has an omnidirectional pattern – meaning the entire red area is equally sensitive to sound.
- Mic 2 has a figure-8 pattern – meaning the two blue areas on the front and back are sensitve, while the sides are ignored.
- Mic 3 has a cardioid pattern – meaning the green area in front of the mic is most sensitive, the sides are less sensitive, and the rear is ignored.



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When to Use Omni's

Because they are so prone to off-axis spill Omnidirectional mics aren't nearly as popular as they were prior to the invention of the cardioid pattern. But by no means does that make them irrelevant. For example.

Here are common situations when they're preferable:

- When recording the sound of the room – such as with room mics for drums.
- When recording a wide sound source – such as an orchestra, choir, or grand piano.
- When recording a moving target – such as an acoustic guitar player who can't sit still.
- When recording in stereo – such as with the common A/B technique.

Compared to cardioid mics, of omnidirectional mics offer the following advantages:

1. Immunity to proximity effect - https://www.youtube.com/watch?v=CGTlPEzaCsE
2. lower self noise
3. a frequency range that typically extends a full octave lower
4. less coloration of off-axis sounds

This last advantage is especially true with small diaphragm omni mics. That is why most precise measurement microphones (like Earthworks mics for example) are small diaphragm omni's.

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When to Use Figure-8

So why exactly would you want a mic that was equally sensitive on both sides? It doesn't seem very useful, does it?

It's common to use figure-8 mics for one of the following 3 reasons:

- for stereo recording
- with ribbon mics
- for maximum isolation of off-axis sounds

For **stereo recording**, figure-8 mics are required to perform both the **Blumlein Pair**, and **Mid/Side** stereo techniques.

- With ribbon mics, the physical make up of the design often requires a figure-8 polar pattern. If you like ribbon mics for their sound, the figure-8 pattern simply comes as part of the package.
- To isolate instruments in close proximity, figure-8 mics are ideal because they completely reject sound from the sides.
- With smart positioning, you can achieve more isolation with a figure-8 mic than with any other polar pattern. One common trick is to place acoustic absorption at the rear end of the mic to block out any unwanted noises.

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When to Use Cardioids

The advantage of using cardioid mics seems simple, It records where you point it, and ignores everything else. Which is why it is the obvious choice for vocal mics.

But here are some less-obvious examples when it's especially useful:

- Miking up a drum kit – With so many instruments so close together, isolation might seem impossible. But it CAN be done, with the right cardioid mics, positioned in the right spots.
- Live performances – On-stage, when sounds are coming at you from all directions, cardioid mics are great maintaining isolation and preventing feedback.
- Untreated rooms – In rooms with poor acoustics, close-miking with cardioid mics can work wonders at minimizing reflected sound.

They might seem ideal in most cases but cardioid mics DO have drawbacks.

- The biggest ones being:
- Off-axis coloration – With most cardioid mics, you see a drop in high frequency sensitivity as sounds move further off-axis. This could be bad, for instance, with an inexperienced singer unconscious of his head movements.
- Proximity effect – A phenomenon exclusive to cardioid mics…proximity effect is a boost in bass frequencies that results from extreme close-miking. Using the same "inexperienced singer" example, you can see how this might also cause problems.
- Plosives - Consonants that cause plosives to be recorded are most often the Pa, Ba, and Fa sound, although other "stop consonants" such as Ta or Ka, can cause the release of air needed to create a plosive.
- Phasing - defined as timing differences when combining identical (or nearly identical) signals. This can be a result of static delay between the signals.

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Different Mics at a Glance

Dynamics

Shure SM58: Produced since 1966 by Shure Incorporated, it has a strong reputation for its durability and sound, and half a century later it is still considered the industry standard for live vocal performance microphones. Like all directional microphones, the SM58 is subject to proximity effect, a low frequency boost when used close to the source. The cardioid response reduces pickup from the side and rear, helping to avoid feedback onstage. Frequency range: 50 - 15,000 Hz.

Shure SM57: The cardioid pickup pattern isolates the source and effectively reduces the recording of background noise. In the studio, the SM57 is ideal for recording drums, guitar, and woodwinds. It has a bright, clean sound and carefully contoured presence rise make it the ideal tool for live performances and studio recordings. Frequency response: 40-15000 Hz.

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Condensers

Neumann KM184: The KM 184 is a transformerless FET (type of transistor) pencil condenser mic, it has a cardioid pickup pattern; other mics in the series provide omni and hyper-cardioid responses. The KM 184 has a Very smooth frequency responses not only for the 0° axis, but also for lateral (off-axis) sound incidence. Frequency response: 20-20000 Hz

Neumann U87Ai: The U 87 Ai is probably Neumann's best known and most widely used studio microphone and is the latest version of the classic. The U 87 is equipped with a large dual-diaphragm capsule with three directional patterns: omnidirectional, cardioid and figure-8. A 10 dB attenuation switch is located on the rear. It enables the microphone to handle sound pressure levels up to 127 dB without distortion. Furthermore, the low frequency response can be reduced to compensate for proximity effect. Frequency range: 20-20000 Hz.

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AKG 414 XLS: The AKG 414 is a classic large-diaphragm condenser microphone, standard equipment in many recording studios around the world, designed in Austria. In music production, it is often used to mic acoustic guitars or pianos, but its most common use is to record vocals. With five polar patterns, bass cut and attenuation options, it's extremely versatile. And it has low self noise. Dynamic range: 140 dB(SPL) max, Frequency range: 20 - 20,000 Hz.

Sennheiser MKE600: The MKE 600 is the ideal video camera/camcorder microphone due to its high directivity, the MKE 600 picks up sounds coming from the direction in which the camera is pointing and effectively attenuates noise coming from the sides and rear. The switchable “Low Cut” filter additionally minimizes wind noise. A blimp (foam windshield/hairy cover combination) reduces wind noise even more effectively, making the MKE-600 the perfect choice for outdoor recording. Because some video cameras/camcorders do not provide phantom power, the MKE 600 can also be battery powered. Frequency range: 40 / 20,000 Hz

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Large Diaphram Options

Audio-Technica AT 2020

Polar Pattern: Cardioid

Freq range: 20 - 20kHz

Price: **€105**

https://www.thomann.de/ie/audio_technica_at_2020_bundle.htm

T.bone SC 450

Polar Pattern: Cardioid

Freq range: 20 - 18kHz

External: -10 dB pad switch

External: low cut switch

Price: **€111**

https://www.thomann.de/ie/the_tbone_sc450_valueset.htm

Behringer B2 Pro Bundle

Polar Pattern: Cardioid, omnidirectional or figure-of-eight

Freq range: 20 - 20kHz

External: -10 dB pad switch

External: low cut switch

Price: **€125**

https://www.thomann.de/ie/behringer_b2_pro_bundle.htm

Rode NT1-A

Polar Pattern: Cardioid, Freq range: 20 - 20kHz

Price: **€175**

https://www.thomann.de/ie/rode_nt1a_complete_vocal_bundle.htm

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3. AUDIO FORMATS & SAMPLE RATES

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Common Audio Formats

Audio files come in all types and sizes. And while we may all be familiar with MP3, what about AAC, FLAC, OGG, or WMA? Why do so many audio standards exist? Is there a best audio format? Which ones are important and which ones can you ignore? Once you realize that all audio formats fall into three major categories. Once you know what the categories mean, you can just pick a format within the category that best suits your needs.

Terms used:

- **Sample Rate/Frequency:** The sampling rate refers to the number of samples of audio recorded every second. It is measured in samples per second or Hertz (abbreviated as Hz or kHz, with one kHz being 1000 Hz). An audio sample is just a number representing the measured acoustic wave value at a specific point in time. Some of the values you might have come across are 8kHz, 44.1kHz, and 48kHz.
- **Sample Depth/ Sample Size:** Measured in bits per sample, the sample depth, (also known as the sample precision or sample size), is the second important property of an audio file or stream, and it represents the level of detail, or "quality" each sample has. What does "quality" mean? For an audio sample, it simply means that the audio sample can represent a higher range of amplitudes.
- **Bit Rate:** Since the sampling rate is measured in samples per second and the sample depth is measured in bits per sample, it is therefore measured in (samples per second) x (bits per sample) = bits per second, abbreviated as bps or kbps. Applications that require high audio quality, like music, usually have a higher bit rate yielding higher quality, or "crisper" audio.

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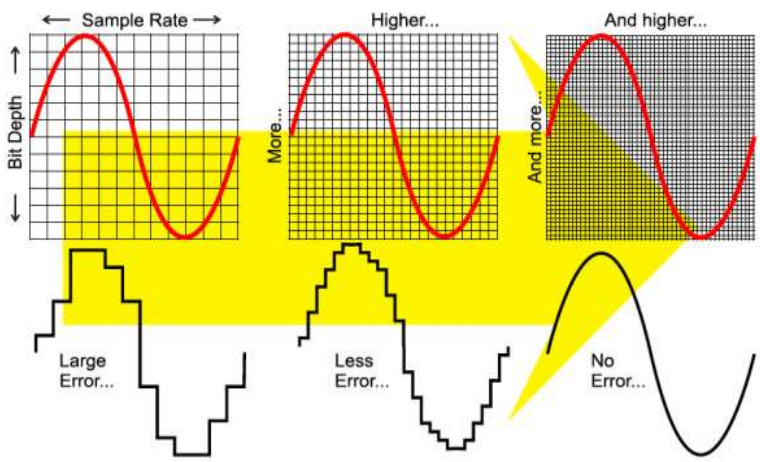

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Uncompressed Audio Formats

Uncompressed audio consists of real sound waves that have been captured and converted to digital format without any further processing. As a result, uncompressed audio files tend to be the most accurate but take up a LOT of disk space---about 34 MB per minute for 24-bit 96KHz stereo.

Audio File Format: PCM

PCM stands for **Pulse-Code Modulation**, a digital representation of raw analog audio signals. Analog sounds exist as waveforms, and in order to convert a waveform into digital bits, the sound must be sampled and recorded at certain intervals (or pulses).

This digital audio format has a "sampling rate" (how often a sample is made) and a "bit depth" (how many bits are used to represent each sample). There is no compression involved. The digital recording is a close-to-exact representation of analog sound.

Audio File Format: WAV

WAV stands for **Waveform Audio File Format**. It's a standard that was developed by Microsoft and IBM back in 1991. A lot of people assume that all WAV files are uncompressed audio files, but that's not exactly true. WAV is actually a Windows container for different audio formats. This means that a WAV file could potentially contain compressed audio, but it's rarely used for that.

Most WAV files contain uncompressed audio in PCM format. The WAV file is just a wrapper for the PCM encoding, making it more suitable for use on Windows systems. However, Mac systems can usually open WAV files without any issues.

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Audio File Format: AIFF

AIFF stands for Audio Interchange File Format. Similar to how Microsoft and IBM developed WAV for Windows, AIFF is a format that was developed by Apple for Mac systems back in 1988.

Also similar to WAV files, AIFF files can contain multiple kinds of audio formats. For example, there is a compressed version called AIFF-C and another version called Apple Loops which is used by GarageBand and Logic Audio. They both use the same AIFF extension.

Most AIFF files contain uncompressed audio in PCM format. The AIFF file is just a wrapper for the PCM encoding, making it more suitable for use on Mac systems. However, Windows systems can usually open AIFF files without any issues.

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Audio Formats With Lossy Compression

Lossy compression is when some data is lost during the compression process---and compression is important because uncompressed audio takes up lots of disk space. In other words, lossy compression means sacrificing sound quality and audio fidelity for smaller file sizes. When it's done poorly, you'll hear artifacts in the audio. But when it's done well, you won't be able to hear the difference.

Audio File Format: MP3

MP3 stands for **MPEG-1 Audio Layer 3**. It was released back in 1993 and exploded in popularity, eventually becoming the most popular audio format in the world for music files. There's a reason why we had "MP3 players" but not "OGG players"!

The main goal of MP3 is three-fold: 1) to drop all the sound data that exists beyond the hearing range of normal people, and 2) to reduce the quality of sounds that aren't easy to hear, then 3) to compress all other audio data as efficiently as possible.

Nearly every digital device in the world with audio playback can read and play MP3 files, whether we're talking PCs, Macs, Androids, iPhones, Smart TVs, or whatever else. When you need universal, MP3 will never let you down.

Audio File Format: AAC

AAC stands for **Advanced Audio Coding**. It was developed in 1997 as the successor to MP3, and while it did catch on as a popular format to use, it never really overtook MP3 as the most popular.

The compression algorithm used by AAC is much more advanced and technical than MP3, so when you compare the same recording in MP3 and AAC formats at the same bitrates, the AAC one will generally have better sound quality.

Even though MP3 is more of a household format, AAC is still widely used today. In fact, it's the standard audio compression method used by YouTube, Android, iOS, iTunes, later Nintendo portables, and later PlayStations.

Audio Formats With Lossless Compression

Opposite to lossy compression is **lossless compression**, which is a method that reduces an audio file's size without ANY loss of data between the source audio file and the compressed audio file.

The downside is that lossless compressed audio files are bigger than lossy compressed audio files---up to 2x to 5x larger for the same source file.

Audio File Format: FLAC

FLAC stands for **Free Lossless Audio Codec**. It has quickly become one of the most popular lossless formats available since its introduction in 2001. FLAC can compress an original source file by up to 60 percent without losing a single bit of data. What's even better is that FLAC is an open-source and royalty-free audio file format, so it doesn't impose any intellectual property constraints.

FLAC is supported by most major programs and devices and is the main alternative to MP3 for music. With it, you basically get the full quality of raw uncompressed audio at half the file size. That's why many see FLAC as the best audio format.

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4. AUDIO INTERFACES

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What Is An Audio Interface?

An audio interface is one of the most important parts to your setup, whether you're building a home studio or portable recording setup. It is the central hub to all of your audio, converting your instrument's sound from analogue to digital for your PC digital audio workstation (DAW) to process. It's important you get the right one to suit your needs.

Interface Size

The First thing to establish when choosing an I/O interface is the environment you are planning to record in. Are you hoping to set up an established project studio/producer setup in the spare room? Maybe you're after a desktop interface you can keep next to your computer. If you need something truly portable, you might be happy with a simple all in one device that's discrete so that you can record at the press of a few buttons.

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The desktop audio interface is the perfect choice for a small project studio. Most desktop interfaces have a good range of inputs and the essential features you'll need to record without going overboard on inputs. Rackmount interfaces are designed to fit into an audio rack so that you can combine lots of outboard gear without taking up too much space. The larger format provides room for more inputs and preamps. Ideal for larger studios.

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Connectivity

Interfaces contain a variety of connections. You need to make sure you can plug your interfaces into your laptop or PC. Simply check what inputs your device has before buying a new audio interface.

- **USB** - The standard connection every PC uses. Most modern wired gadgets connect to your computer this way, including phones, cameras and tablets. USB 2.0 has been out for a long time and has a relatively average connection speed - otherwise known as latency. However, new computers and the latest interfaces have USB 3.0 (USB-C) connections that offers wider bandwidth (Amount of information that can be transferred).
- **Thunderbolt** - Created by Intel, Thunderbolt is the fastest easily obtainable audio interface connection. There are now a number of Thunderbolt Interfaces out on the market, which offer ultra-low latency and unparalleled performance. All the latest Macs come equipped with Thunderbolt inputs, but if you've got a Windows PC you might need to get an input specifically installed. This is the successor to Firewire.
- **PCI Express** - An internal card-based computer connection platform primarily found in desktop computers. Since these cards are plugged directly into the PC motherboard, they require an available PCIe slot for installation, which some computers may lack. The PCIe connection provides high data bandwidth and low latency, allowing audio interfaces that use it the ability to handle many simultaneous inputs and outputs.
- **Ethernet** - Extremely high-end. It's not needed for home use as the quality doesn't justify the cost. But it's the best option for professional studios and broadcast facilities that have tens of thousands to spend on recording equipment.

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Inputs and Outputs

Sound can be captured in a variety of different ways so it's important to work out the quantity and type of inputs you'll need. Recording instruments or vocals through a microphone requires an XLR connection.
These carry analogue signals. They're easily recognisable because they are the largest slots you'll see on the interface. An interface has TRS outputs so you can plug your headphones and speakers in to listen back to what you're recording.

If you're a singer-songwriter you may only need a small interface with a couple of inputs for overdubbing. However, if you're looking to record a 5 piece band all at the same time, you'll need a mixture of inputs; including a number of microphone inputs for the drums alone.

Analogue I/O

- **Jack** - Inputs used for guitars, bass, keyboards, synths and external gear. Stereo instruments like Keyboards will need 2 jack inputs for the right and left channels. Outputs are used for connection to monitor speakers and any analogue effects and other gear that you want to feed through an effects channel (bus).
- **XLR** - Balanced Microphone inputs. XLR mic inputs are combined with a microphone preamp that boosts the signal to a usable level. Each preamp comes with a gain knob so you can set the level of boost on the signal.
- **Combo Jack XLR Inputs** - Audio interfaces usually have a combination of jack and XLR so that you have the option of plugging in easily without having to reach to the back of the unit.
- **3.5mm Jack** - Allows you to hear what you're recording through headphones or speakers. If you want an accurate representation of the sound you're producing, get some good full range monitors.

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Digital I/O

- **S/PDIF (Sony/Philips Digital Interface)** - This is the main input/output for any digital sources. The digital input saves you from any noise that might come into the signal through an analogue input. S/PDIF comes in 2 different formats; Coaxial or Optical. Make sure that any devices plugged in have the right connection.
- **MIDI** - Used to plug in peripherals that can control the software in your computer. For example, a keyboard can be hooked up so to play software instruments hosted on the computer. Some modern dedicated controller peripherals have USB connections that plug directly into your computer. However with most keyboards, MIDI will be your only option.
- **ADAT** - An optical connection that allows you to transfer digital audio between different equipment. It can transfer up to 8 channels at 48 kHz/24bit.

Quality

"YOUR SIGNAL CHAIN IS ONLY AS STRONG AS YOUR WEAKEST LINK!"

There are a number of key elements in an audio interface that will determine the overall quality of your system such as the cable and interface.

You may have the most expensive microphone that captures sound in incredible detail , only for it to run through a bad cable or into a poor interface. All that gorgeous detail and fidelity can be lost or drowned out by background noise.

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Summary

In an ideal world, we would all love to get the biggest and best audio interface out there for our studio. Unfortunately, high-end recording gear can easily run into the thousands, and won't bring you the leaps in quality you might expect.

Generally, within the various price bands, the more you spend the higher the quality of interface. However, there is a delicate balance between better sound and number of inputs.

The best audio interface for you is one that has all the functionality you need within your budget. Choosing the right interface for you may seem daunting at first, but by carefully considering your requirements, you can very quickly narrow down your choices.

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Interface Examples

Presonus AudioBox USB 96 25th Annivesary Edition

2 Mic / instrument inputs: XLR combo socket

2 Line outputs: 6.3 mm jack balanced

MIDI input and output

Stereo headphone output: 6.3 mm jack

USB 2.0 port

Phantom power + 48V can be activated

Price: €88

Tascam US-2x2HR

2 Microphone inputs with Ultra-HDDA microphone preamps with low Inherent-noise 24-Bit / 192 kHz

2 Line outputs: 6.3 mm jack, balanced

Stereo headphone output: 6.3 mm jack

MIDI I/O For PC / MAC

USB bus powered

Switchable +48 V phantom power

Price: €155

Focusrite Scarlett 4i4 3rd Gen

2 Mic / instrument / line inputs: XLR / 6.3mm jack combo balanced 24-Bit / 192 kHz

4 Line outputs: 6.3 mm jack, balanced

Stereo headphone output: 6.3 mm jack

MIDI I/O For PC / MAC

USB-C bus powered

+48 V phantom power

Price: €195

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5. RECORDING & EDITING SOFTWARE (REAPER).

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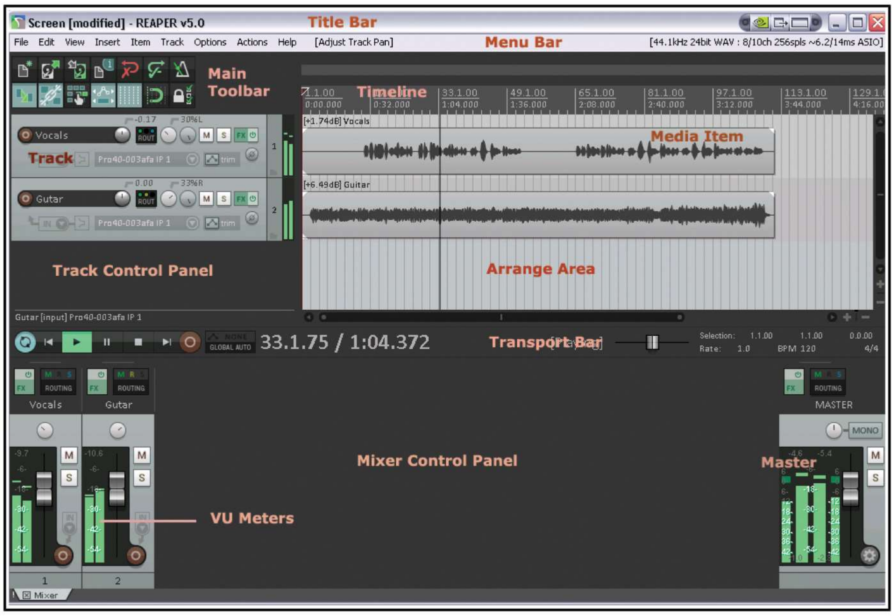
The Interface



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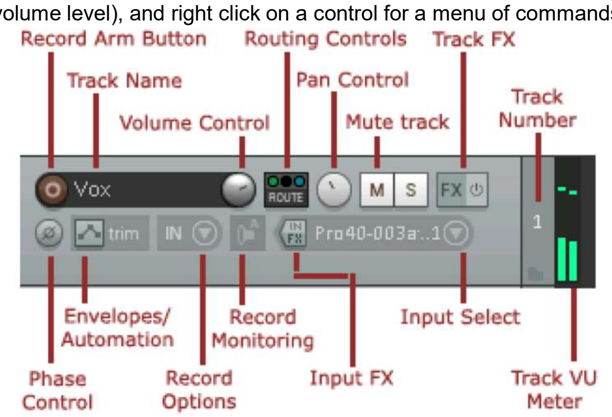
The Track and Track Controls

If you've used other DAW software before you'll probably want to get to grips with REAPER's track controls as soon as possible. This illustration shows the most commonly used of these. The exact position of some of them will vary with track control panel width. You can hover your mouse over any control for a tooltip. In most cases you click on a
control to use it (for example, click on Mute button to toggle mute status of any track, click and drag on Volume Control to adjust the volume level), and right click on a control for a menu of commands, options and/or settings.



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Track Controls

Panning: Click and hold down the left mouse button over the rotary fader, then drag in the direction required, Releasing the mouse when finished. For rotary faders, drag up to rotate clockwise, down for anti-clockwise. Hold **Cmd** while doing this for more subtle adjustments.

Volume: The rotary fader shown here is used to adjust a track's **volume**. With some layouts, this will be shown as a horizontal fader.

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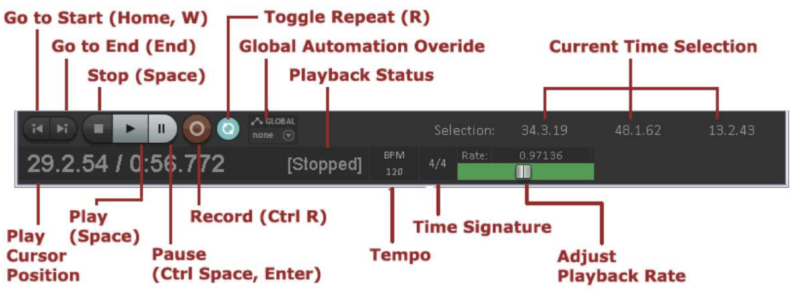
The Transport Bar

If the Transport Bar is not visible in your REAPER window, use the **View, Transport** command to display it, or use the keyboard shortcut **Cmd Alt T** to do this. The Transport Bar might appear as shown here, as a floating window (undocked) inside REAPER, or it might be docked just below
arrange view. At this stage, the exact position of the Transport Bar is not important. The main Transport Controls are shown below, though your exact layout might not be identical: **whether docked or not, the controls will appear in a single row if the window is wide enough.**



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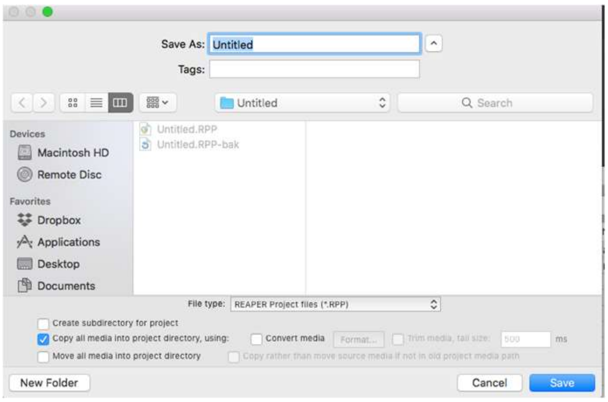
Saving Your Project

1. To save your file in Reaper go to the File menu and select "Save file as". This will open a pop up dialogue where you can enter your file name and select a location in which to save your project.
2. When giving your project a name it is recommended that you follow it with a version number for sake of easy identification i.e. "Kieran's Project v1".
3. Once you set a location to store your file there is one more setting that should be checked. In the bottom portion of the box you have three optional check boxes. Please check the box next to "Copy all media into project directory, using:", this is especially important if you will be editing the project file on multiple computers and using a portable drive or USB stick to store your files. Otherwise you **WILL** be missing some of your audio assets when your open the project.



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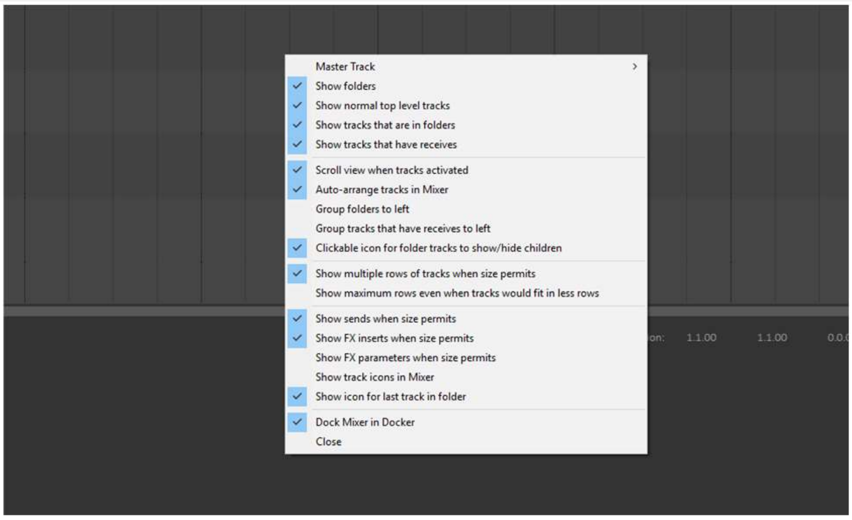
Creating Markers & Housekeeping

Once we open Reaper we will make a number of minor changes to the settings for the purposes of this project.

1. Right click on the mixer pane, make sure that “Clickable icon for folder tracks to show/hide children” is checked



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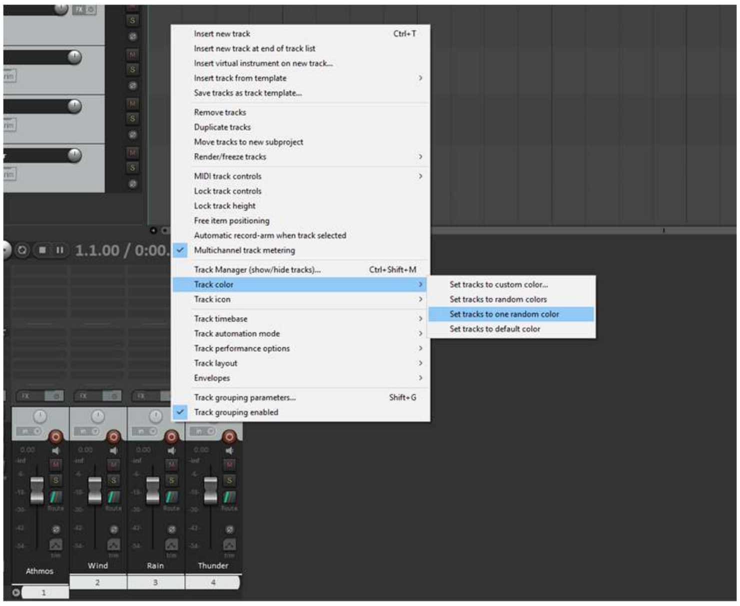

2. After we create some tracks and group them we are going to add a colour to them to make them more easily identifiable.

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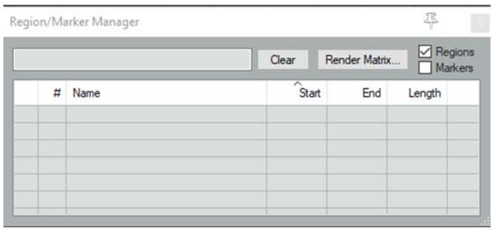
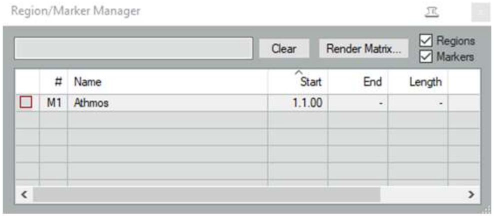
3. Move the playhead to the beginning of your first song and create a marker by pressing the “M” key. N.B. To remove a marker hold “Option/Alt” and click the marker.
4. Then we are going to open the Region/Marker Manager by pressing “Ctrl+Alt+Shift+R” together.
5. Once this is opened make sure the check box next to “Markers” is selected. Then click the pin in the top right of the box. This pins the box so we can find it easily.





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Creating Regions

Creating Regions

1. Before you start, you may wish to turn off snapping on the toolbar on the top right of the screen, this will allow you to select your regions more freely. To create a Region you need a start and an end point. Move your cursor to the timeline where you want your region to begin and then drag it right to where you want it to end. The highlighted section will be your region.

2. Right click the selected area on the timeline and select “Create region from selection” on the menu which drops down. You could also use the shortcut “Shift+R”.

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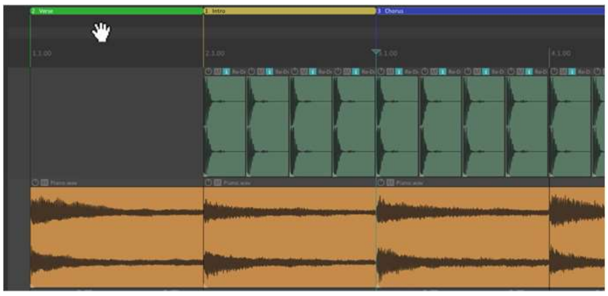
3. By using regions it allows us to move around sections of our project more efficiently and to arrange sections easily. When you move a region you move all audio within that region also.



4. Here you can see that by simply dragging our sections we can entirely rearrange a piece in seconds.
5. You can also edit the colour of each region to more easily identify them by right clicking and selecting "Edit region".

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Automation with Envelopes

Automation is used to ensure that when your tracks are played, recorded changes in such things as volume level or panning can be recalled with the music in real time. At its simplest, for example, automation can lift the volume of a lead instrument during a particular break or passage, or add a little presence or warmth to the odd phrase here or there on a vocal track, to make it stand out in the mix a touch more.

An example of a simple envelope (for Volume) is shown here. The envelope is in this case displayed below the media item. In the TCP you can see an envelope panel with its own envelope controls, just below the track controls. When the track is played, the volume of the track will rise and fall, following the shape of the envelope.

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How to Use a Parametric Equalizer

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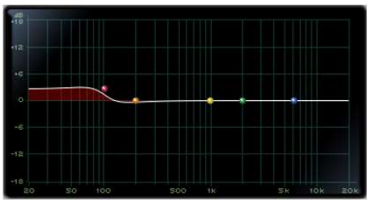
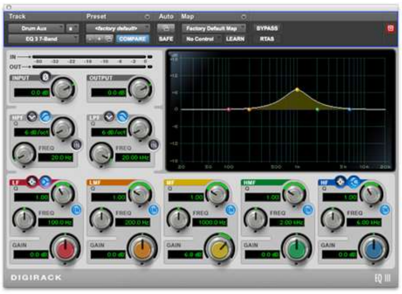
When you're setting up a session, adding an EQ as the first insert is almost as essential as creating the tracks themselves.

There are several controls on the EQ and some will see more use than others, we'll take a look at the less frequently used ones to begin and then move onto the others.



- **In/on:** the In button turns that particular EQ control on or off. If you equalize a frequency and then decide you don't want to keep the change, you can just turn off that EQ band until you need it, instead of having to zero out the settings.
- **Shelf/Notch:** the two buttons next to the EQ band name determine the shape of your EQ when it is at one end or the other of the frequency spectrum. Those in the middle are notch, meaning they affect a set range of frequencies, but these end bands can be shelved which means they are affected from the bottom of the frequency spectrum (for the LF) up to the set frequency, or the top of the spectrum for the HF.



Gain

Once you've pushed a band in, the first control you should tweak is the Gain control. The EQ will have no effect without some gain reduction or addition, no matter what you do with the other controls. Gain determines how much of a certain frequency is added or removed. Gain is the vertical axis on the EQ graph, and the taller it is, the more of that frequency is being added:

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Q

Q determines how wide or narrow the EQ band is. A setting of 0 will pretty well encompass the entire spectrum (depending on your gain) while a setting of 10 will only affect a very small range of frequencies. Here's a Q that's fairly average, though a little on the narrow side:

Frequency

The third control is Frequency. This determines which frequency the band affects, or in most cases where the Q determines that a range of frequencies will be affected, where the center of the frequency range is.

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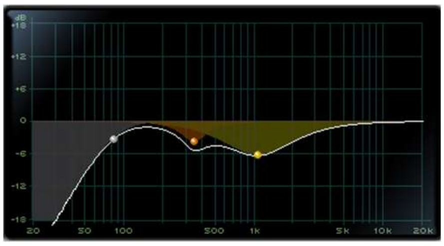
Subtractive EQ is the Best EQ

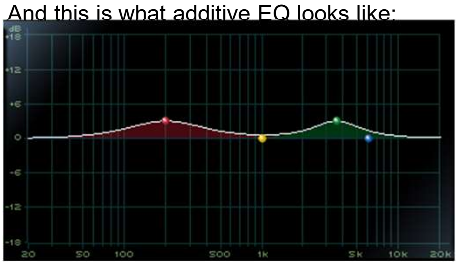
As we just discussed, you can either boost a frequency or attenuate it. However, just because you can boost doesn't mean you necessarily should: it's better to pull frequencies down. When you increase a frequency, the plug-in has to create extra sounds that weren't there before. When you attenuate, you're just reducing part of the existing sound, so it stays more natural and realistic.

So what do you do if you want to get a beefier bottom-end? Simple - pull down the high-end!

Here's a picture of what subtractive EQ looks like:



And this is what additive EQ looks like:



Of course, you can boost if you want to, but it's a good idea to try the subtractive approach first.

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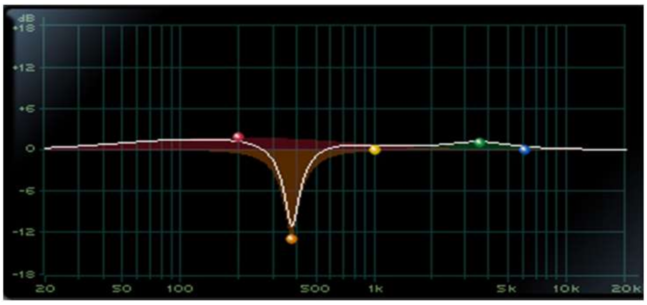
Cut Narrow, Boost Wide!

While we're on the topic, there are a few best practices for cutting and boosting frequencies. When you're cutting a frequency, it's best to make it narrow (a higher Q) and a bit deeper, whereas if you're boosting, it's better for it to be wider (a low Q) but shallower (in other words, use gain sparingly).

This is not a hard and fast rule. If you're recording at home, or have a less-than-perfect take in the studio, you may find yourself cutting wide fairly regularly. And if you just need a sound to poke through in a small range, you might introduce a narrow (though still shallow) EQ.

You can get an idea of how it's done by looking at this image:



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Low and Hi Pass Filters

I find one of the reasons I most frequently use EQ is to slap a low or hi pass filter on.

A low pass filter attenuates the high frequencies, while the high pass filter does the same for the low frequencies. Usually it's attenuation that's happening, though you might find a slight LPF boost on drum overheads, for example, gives things a bit more sparkle.

Frequently when you mix a session, you may put a HPF on every track other than the kick drum and bass guitar (unless you have other instruments in the session that are there to be bass instruments). Low frequencies get muddy ridiculously quickly, so it's important to be ruthless to keep your studio sound chrisp. It's worst in rock and metal, so do some tests before taking what is discussed here as set in stone.

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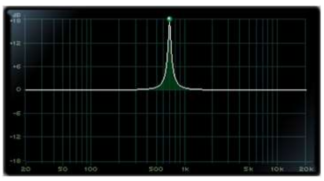
Cutting Holes

Even when you're not controlling low frequency build-up, you'll need to cut holes in certain instruments. It's important to let an instrument dominate it's primary frequency range, so you should cut holes in your instruments that will compliment each other. For instance, the human voice is (generally) strongest in 3.5kHz, so if you find another instrument is competing with the vocalist in that range, pull them down there like so:

This ties into planning your session and arrangement. Find instruments to fill up each major block along the spectrum, and then make sure they are the strongest in their ranges with the help of subtractive EQ.

Finding Problem Frequencies



If you find the snare is too boxy or the guitars are too jangly, you'll be using EQ to remove "problem" frequencies and this trick will help you identify the problem and fix it.

Give one of your bands a high Q (10 is not too high in this case) and raise the gain as high as it goes.

Now you need to perform a "sweep" along the spectrum until the problem sound becomes really prominent. When you find the frequency where the problem is at it's worst, reduce the gain and change the Q until you've controlled it. It's a fairly simple trick, but you'd be surprised that so many people attempt to fix a problem with EQ before they've located the frequency range where it's occurring.

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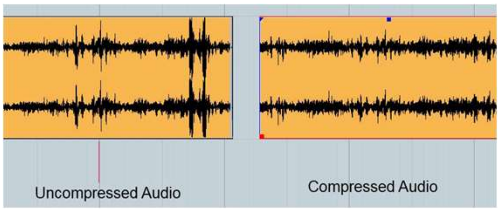
What is an Audio Compressor?

Audio compression reduces the dynamic range of your recording by bringing down the level of the loudest parts, meaning the loud and quiet parts are now closer together in volume and the natural volume variations are less obvious.

The audio compressor unit can then boost the overall level of this compressed signal. So the end result is that the quieter parts sound like they've been boosted in volume to be closer to the louder parts.

The dynamic volume changes of a recording are now under more control, and a knock-on effect is that the overall level of the compressed recording can be increased inside your mix. The recording will also sit inside your whole mix much more easily.



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Compression Controls

The compression device itself has many different controls that can have an effect on the sound you're processing. Lets run through the main controls that are commonly found.

Input Gain

- This controls the level of the signal going into the audio compressor.

Threshold

- Compression brings down the overall level of the loudest parts of your recording. But how does the compressor know which part of the signal is 'loud' and which part of the signal to compress? By setting the threshold.

- The threshold sets the level at which the compressor kicks-in and starts changing the dynamics of the recording. So for example, if you set your threshold at -20 dB, everything below this level will not be affected by the compressor. But everything louder than this level (-20 dB) will be compressed.

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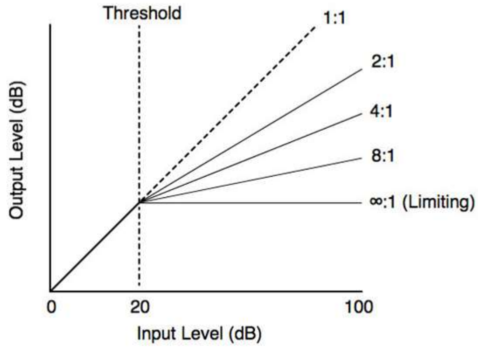
Ratio

- How much will the signal be compressed once it's gone over this threshold level? This is controlled with the ratio. The higher the ratio, the more compression there is.
- The easiest way to show you how the ratio works is by showing you some numbers.
- If the ratio is 1:1, there is no compression at all.
- If the ratio is set at 2:1, for every 2 dB of sound that goes over the threshold, you get 1 dB of output above the threshold. So if the signal goes over the threshold by 10 dB, the compressor reduces this signal so it's now 5 dB over the threshold.
- If the ratio goes up to 8:1, for every 8 dB of sound over the threshold you would get 1dB of output above the threshold. So if the signal goes over the threshold by 16 dB, the compressor reduces this so only 2 dB goes over the threshold.



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Attack

- This is the time that the compressor takes to act on the input, once the sound level has gone over the threshold level. It's usually measured in milliseconds (ms).

Release

- This is the time that the compressor takes to let the signal return to normal once it has fallen below the threshold level. Again, usually measured in ms.

Output Gain (Make-up Gain)

- If the audio signal has been compressed, the overall level of the signal will be reduced. Increasing the output gain raises the level coming out of the compressor, so the volume can be more easily matched to the levels of the rest of your tracks in your mix.

Knee

- Soft-knee compression is gentler on the sound as it goes through the audio compressor - the change from uncompressed to compressed sound is smoother. Hard-knee compression is a more immediate and obvious effect.

Compressors that are specifically designed to offer very high-ratio compression are often called limiters, so if you find that your compressor simply can't muster a high enough ratio to do a particular job, don't be afraid to try a limiter instead.

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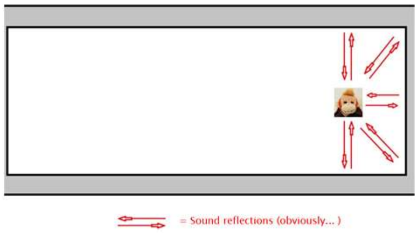
What Is Reverb?

Reverb is created when a sound occurs in a space, sending sound waves out in all directions. These waves reflect off surfaces in the space, decaying in amplitude until the reflections eventually die off. Without extensive sound-proofing, most spaces will produce many closely spaced reflections, which reach the listener shortly after the initial dry sound. We hear this series of reflections as a single, continuous sound, which we call “reverb.”

To explain how this works, imagine a rectangular shaped room with hard walls, similar to the Diagram on the right, with a willing subject ready to make some noise within. When our subject makes a noise, it will travel in all directions from the source at the same speed. When it does, it reaches the walls closest to it first, and is reflected back to the listener the soonest.

These are known as early reflections. The delay time between the source sound and these early reflections is known as pre-delay, and is the main factor that dictates how ‘big’ a reverb is perceived to be.

Early Reflections



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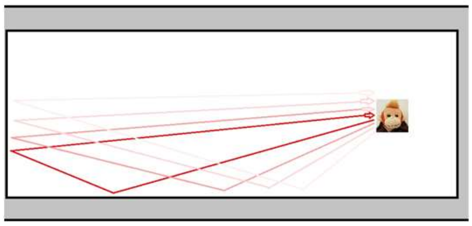
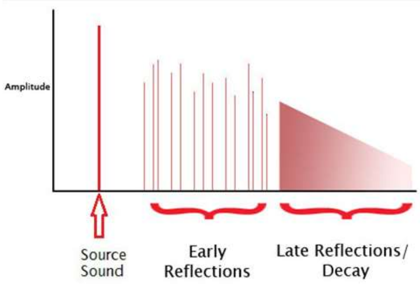
The longer the delay, the further the sound has to travel, so the bigger the space must be. After this phase, the reflections from elsewhere in the space (of which there are far more), which have been bouncing around the other walls and hard surfaces, start to return to the listener.

The shorter the distance these have travelled, the sooner they will arrive, and the louder they will be. Conversely, the further they have travelled, the later and quieter they will be.

Later Reflections and Decay





As these many, many reflections are arriving at fractionally different times, the effect is of a gradually decaying sound, known as reverberation.

The length and character of this decay is affected by many factors, such as room size, shape, number of hard surfaces, and the materials that these are made of. So a big Cathedral should have a long pre-delay, and a long decay, due to its size, shape and construction, whilst, say, your living room will most likely be a far shorter delay and decay.

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What are the types of reverb sounds?

When we say "types of reverb sounds," we're talking about the methods through which reverb is created. The following examples show the most common methods in action. No direct signal is included, so you can focus on the reverb.

Hall reverb

Hall reverb results from the unique physical characteristics of concert halls, which are typically large spaces acoustically designed for a long, smooth decay.

Room reverb

Room reverb results from the unique physical characteristics of smaller rooms like studios or living rooms, typically with shorter decay times and closer reflections.

Chamber reverb

Chamber reverb results from the unique physical characteristics of reverb chambers, which are reflective spaces such as corridors or stairwells designed to house a speaker and microphone configuration for triggering and recording the reverb.

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Plate reverb

Plate reverb results from a vibrating metal plate. In a real plate reverb, a large sheet of metal is suspended in an enclosure. Multiple transducers—a small driver and at least one small contact microphone or pickup—are attached to the plate. Dry signal is sent from a console or audio interface to the driver, which causes the plate to vibrate. The contact microphone picks up these vibrations and outputs them for use in the mixing system. The larger the plate and the further apart the transducers, the longer the reverb time.

Spring reverb

Spring reverb results from small vibrating springs. Like plate reverbs, spring reverb units rely on vibrations to create reverb. Dry signal is routed to a transducer, which is attached to one end of multiple springs. Signal passing through that transducer causes the springs to vibrate. Those vibrations are picked up by another transducer on the other end of the springs. The longer the springs, the longer the reverb time.

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General tips for using reverb

Don’t just treat reverb as an ‘effect’ that you’re adding to a sound. You’re actually introducing a brand-new signal in the mix, one that should play well with others and be treated as its own mix element.

The standard way to practice this approach is to use return/auxiliary channels to add reverb, with the dry/wet balance at 100% wet. This keeps all of your dry and wet signals separated, which not only allows us to more intentionally process our dry sounds, but also to independently process our reverb signal. You can always use insert reverb as a sound design tool, but it’s best to separate the dry and wet signals in any standard reverb application.

Like any other mix element, you’ll want to EQ your reverb signal, mostly to attenuate frequencies under about 200 Hz and above about 8 kHz. If your reverb has too much energy in the lows, it can smear your low end and prevent the clarity you need there. Too much energy in the highs, and reverb can make it difficult for vocals and lead instruments to cut through the mix.

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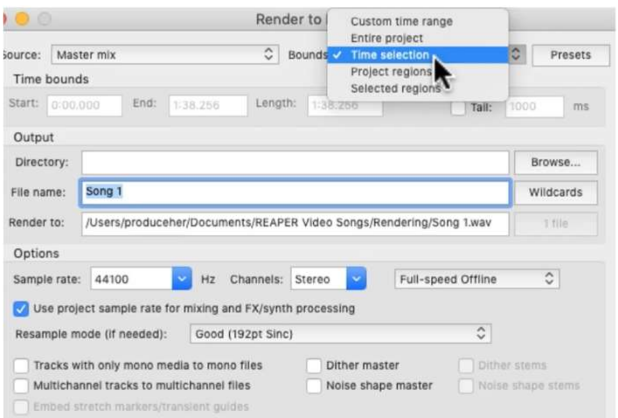
Rendering Audio

1. To Render (Also called Bouncing) your audio files from Reaper you can select “Render” from the file menu or use the shortcut “Ctrl+option+R/Ctrl+Alt+R”.
2. You will then be greeted by a pop up dialogue which will give you multiple selections.
3. Many of these settings will be left as they are but we will make changes to a small few to speed up the whole process.
4. Source: For the sake of ease this will remain the same.
5. Bounds: This we will change to “Project Regions” meaning that it will render each one of our regions to its own individual audio file.
6. Directory: Browse to the folder that contains all your samples and effects, here you will create an new folder to store all your newly rendered files.



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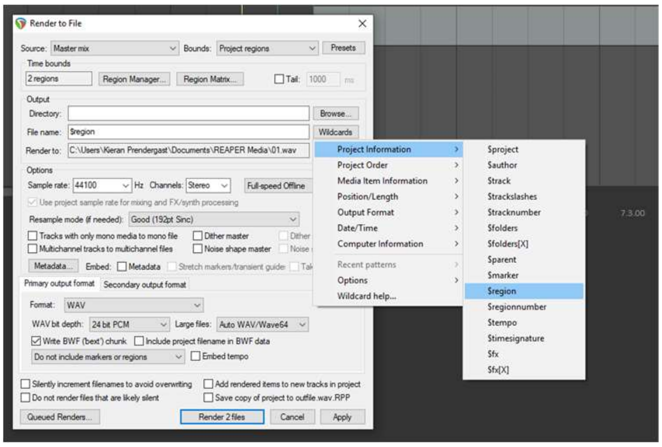
7. File Name: To automatically label all our files reaper allows us to assign the region names to each file as it is bounced. This can be done by clicking the “Wildcard” button, selection “Project Information”, then selecting “$region”.

8. Render to: This is where your final files will be rendered and how many will be rendered.

9. Sample Rate: This is the sample rate your files will be rendered to, here you can see that it will be 44100 samples per second which is equivalent to CD quality.

Moving down the box a little.

10. Format: This will determine the file type used for the final file output, generally .WAV files will use the least amount of compression and maintain the overall quality well as well as being very compatible across all platforms. For use on Apple .Aiff can be used and does not compress the file size heavily.



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6. STEREO MIC TECHNIQUES

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1. A/B Stereo Recording

Mics used: Two omnidirectional mics, usually small diaphragm condensers

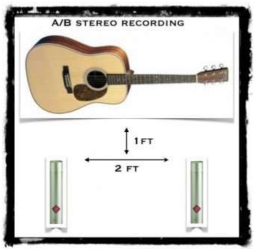
Positioning: Point both mics toward the instrument, at a distance of a foot, and spaced two feet apart.

When experimenting with this technique, try making adjustments to both the distance of the mics from the instrument, as well as the distance of the mics from each other.

How to mix the signals: The mono signals from each microphone are assigned to the left and right channels of a stereo track to create a sense of width in the recording.

How it should sound: The stereo image is created because of the time of arrival at each microphone is slightly staggered. The frequency balance is different as well, which will provide an added level of stereo width.

The downside of A/B stereo recording is that because of the timing offset between each microphone, you will be likely to have issues with phase cancellation when combining the stereo signal to mono.



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2. X/Y Stereo Recording

Mics used: Two directional mics, usually small diaphragm condensers.

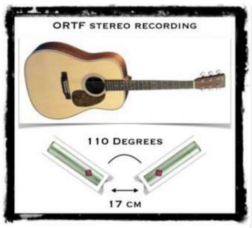
Positioning: at an angle between 90-135 degrees so that their capsules coincide at a single point. The wider the angle, the wider the stereo image.

How to Mix the Signals: (same as A/B stereo recording)

How it should sound: Compared to A/B stereo recording, this technique will have *less* of a stereo effect. The reason is that since both microphones are positioned at the same point in space, there will be no differences in timing.

The entire stereo effect will be created from the differences in frequency balance. The upside to this is there are no issues with mono phase cancellation either.

3. ORTF Stereo Recording



Mics used: Two directional mics, usually small diaphragm condensers

Positioning: Spread outward at an angle of about 110 degrees, with the capsules spaced 17cm apart.

How to mix the signals: (same as A/B stereo recording)

How it should sound: The technique is basically a combination of the previous two. The microphones are physical spaced apart, like with A/B recording, which will yield a wider stereo image. Then it uses directional mics, like with X/Y recording, so it should pick up less of the ambient room sound.

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4. Blumlein Pair

Mics used: two figure 8 (bi directional) mics.

Positioning: (same as X/Y technique).

How to mix the signals: (same as X/Y technique).

How it should sound: Compared to the X/Y technique, the Blumlein Pair technique captures a greater portion of the room sound and adds a bit more ambience to the stereo image, thanks to the use of the figure 8 mics.

5. Mid/Side Stereo Recording

Mics used: One small diaphragm condenser mic – either cardioid OR omnidirectional. One large diaphragm condenser mic – MUST BE FIGURE 8

Positioning: The figure 8 mic is placed sideways at a 90 degree angle from instrument. This mic will record sound on both sides and will function as the "side" in the term mid/side.

The other mic is positioned on top or below the figure 8 mic, and is pointed directly toward the instrument. It will function as the "mid" in the term mid/side.

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How to mix the signals: This part is complicated. Follow the steps carefully.

1. Duplicate the "side" channel.
2. Reverse the polarity of the duplicated channel.
3. Combine the two side channels onto one stereo track.
4. Mix in the mid channel with stereo side channels to adjust the width. The greater level of the sides compared to the mid, the greater the stereo width.

How it should sound: Mid side recording may be complicated, but it offers all the advantages of the other 3 techniques, without the downsides.

- It offers the added stereo width of the A/B technique
- It offers the mono compatibility of the X/Y technique
- And it allows you (if you want) to increase the room ambiance to something resembling the Blumlein Pair technique.

Eventually engineers designed new **cardioid capsules** that were essentially hybrids of the original two designs.

PRACTICAL EXERCISE

WHAT DO I NEED?

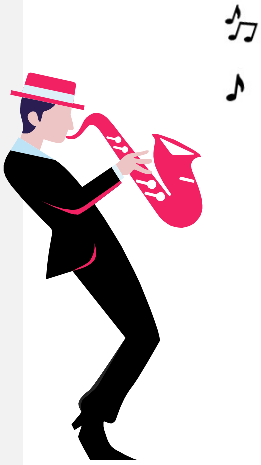
- ✓ A selection of microphones with different polar patterns, two of each type. Two microphone stands, cables, an audio interface, and computer with Reaper installed.

You have 1 hour to complete the task.

OBJECTIVES

01 To identify in which processes used to setup different stereo microphone techniques.

02 To identify the differences in how they sound, and how adjusting the position may help improve quality of the sound.



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7. USEFUL KNOWLEDGE

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AUDIO CONNECTOR TYPES

Signal Cables

RCA Phono: Mostly used for audio Signals but also used to carry analogue video. Phonos use an unbalanced mono signal.

6.35mm Mono Jack: Used for carrying mono signals from instruments to amp or D.I. Box. Also available in 3.5mm (These are sometimes used on mics for radio mic packs.

6.35mm Balanced Jack: Used for carrying a balanced signal, also called TRS Jack (Tip, Ring, Sleeve) Carries signals from outboard equipment to mixing desk. Also available in 3.5mm.

3 Pin XLR: Most commonly used for audio Signals but also used to carry DMX. XLR is a balanced signal cable, good for noise rejection over long distances.

Speaker and Power

NL4 Speakon: 4 pin version of speakon connector used carry power from the amplifier to the speakers.

NL8 Speakon: 8 pin version of speakon connector used carry power from the amplifier to the speakers. It's a little chunkier than an NL4.

EP6 Connector: 6 pin connector used carry power from the amplifier to the speakers.

Powercon: Often used to power active speakers these connectors are now also commonplace on many moving lights and automated fixtures.

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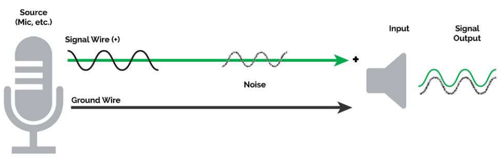
AUDIO CABLES – BALANCED vs UNBALANCED

Unbalanced cables

- The cables themselves consist of two wires inside the plastic casing: a signal wire and a ground wire. The signal wire in the center of the cable passes the audio signal through, while the surrounding ground wire shields the main signal wire from external electronic interference from devices such as lights, televisions, radios and transformers.
- A cable is considered "unbalanced" when it takes the audio signal from a piece of equipment you're using (such as an instrument or stereo system) and passes it straight through to a mixer or other capture/receiver device without manipulation. Leaving the audio untouched makes things simple, but it also means that sometimes the audio can become distorted.



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Unbalanced cables commonly use one of two different types of connectors:

- Standard TS (tip-sleeve) ¼” (6.5mm) cable connectors, the same sort that might be used to connect a guitar to an amplifier.

- RCA cable connectors, which are the red and black tips often used in stereo setups such as surround sound systems, turn tables and older audio systems.

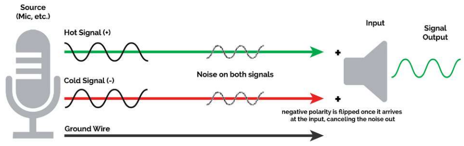
Balanced cables

- The structure of a balanced audio cable is similar to an unbalanced cable — with one addition. A balanced audio cable has a ground wire, but it also carries two copies of the same incoming audio signal, sometimes referred to as a hot (positive) and cold (negative) signal.

What’s the difference between the hot and cold signals?

- The two signals are reversed in polarity, so as they travel down the cable, they cancel each other out. (Think of how adding positive and negative numbers of equal value amounts to zero.)

- Once the hot & cold signal get to the other end of the cable, the polarity of the cold signal is flipped, so that both signals are in phase, and in sync. If the cable picks up noise along the way, the noise added to both of those cables is not reversed in polarity. So when the cold signal flips in polarity to match the polarity of the hot signal, the noise carried along the cold signal cancels out with noise in the hot signal



Balanced cables also have two common typical connector tips:

- TRS (tip-ring-sleeve) cable connectors, such as those used for headphone jacks.

- XLR cables, which are connectors you would typically find on a microphone.

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- The biggest downside to unbalanced cables is the distortion. The ground wire meant to protect the unbalanced cable can, at longer lengths of cable, actually act as an antenna or amplifier for surrounding noise. The hum from a nearby television or audio system could be picked up by the cable heard in your audio capture. As a general rule, this effect is only noticeable with cables beyond 20 feet in length.
- With balanced cables, you don’t get the same distortions, so they can be much longer without any detriment to your sound quality. That said, when the cord length is under 10 feet, unbalanced cables actually have a stronger signal than balanced cables. This is because at this length, any distortion is unlikely, and the simplicity of unbalanced cables can work wonders when there’s no detriment coming from potential distortion.

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SHELS
MUSIC

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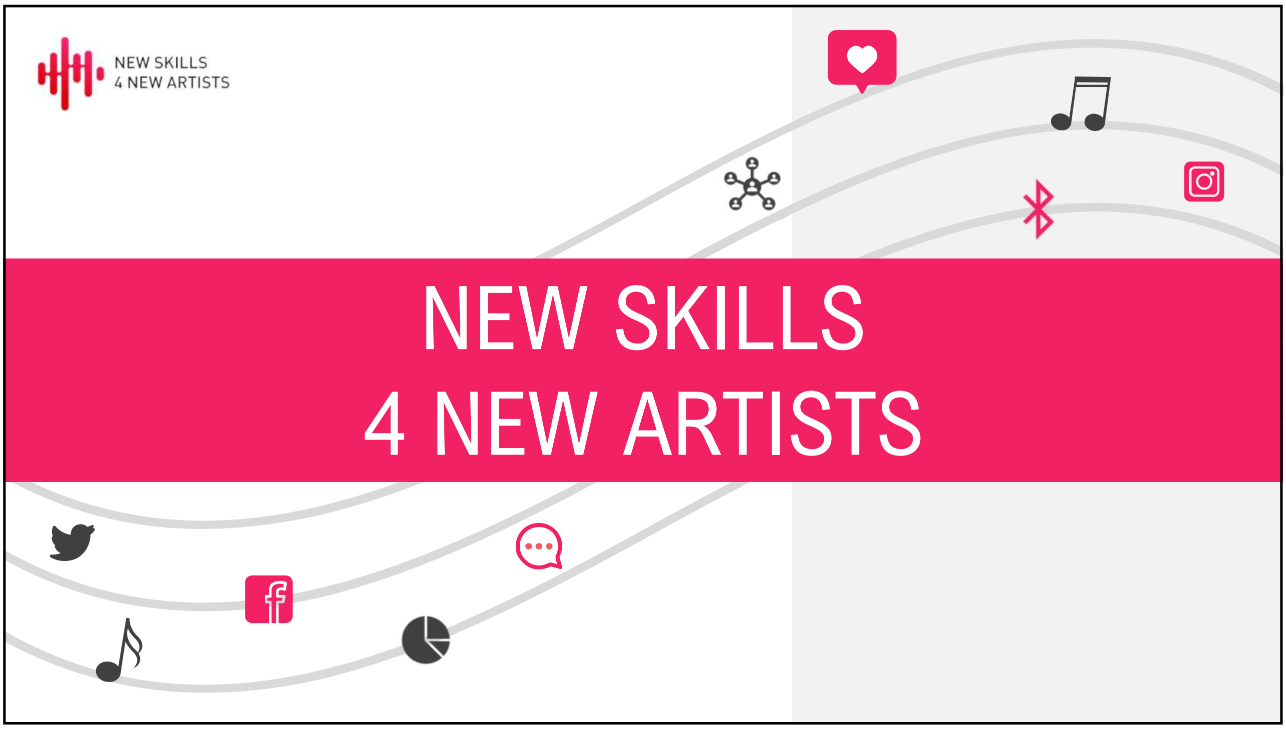
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4 NEW ARTISTS

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MODULE 3

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INTRODUCTION TO TECHNOLOGY (LIGHTING)

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UNITS

NEW SKILLS 4 NEW ARTISTS

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1. THE LIGHTING CHAIN

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The Lighting Chain

- A lighting control console (also called lighting desk) is an electronic device used in theatrical lighting design to control multiple fixtures at once. They are used throughout the entertainment industry and are normally placed at the Front of House (FOH) position or in a control booth.
- All lighting control consoles can control dimmers, which control the intensity of the lights. Some consoles can also interface with other electronic performance hardware (i.e. sound boards, projectors, media servers, automated winches and motors, etc.) to improve synchronization or unify their control.
- Lighting consoles communicate with the dimmers and other devices in the lighting system via an electronic control protocol (Language). The most common protocol used in the entertainment industry today is DMX512, although other protocols such as Art-net and DMX-512-A are evolving to meet the demands of ever increasing device sophistication.

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Types of Console

The Two Scene Preset Desk: Preset boards are the most basic lighting consoles and also the most prevalent in smaller installations. They consist of two or more identical fader banks, called scenes. The faders (sliders) on these scenes can be manually adjusted.

Each scene has the same number of channels, which control the same dimmers. The operator can build a scene offline or in "blind", a cross-fader or submaster is used to selectively mix or fade between the different scenes. You can leapfrog from scene to scene through a show.

This is rather labor intensive for the operator and involves using a set of notes that have been prepared during the technical rehearsal. Preset boards generally control only conventional lights.

Submaster Playbacks: To make life a little easier on the operator these desks allow you to setup a full scene similarly to the two scene preset desk, however, the scene can then be recorded to one single fader.

This allows you to program multiple sub masters (Subs) so that they can be accessed more rapidly during the course of a show. By using this method the operator dramatically cuts down on the amount of paperwork required to run a show.

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Memory consoles: This type of controller has almost completely replaced preset consoles as controllers of choice.

Memory consoles are preferable in productions where scenes do not change from show to show, such as a theatre production, because scenes are designed and digitally recorded, so there is less room for human error, and less time between lighting cues is required to produce the same result.

They also allow for lighting cues to contain larger channel counts due to the same time saving gained from not physically moving individual channel faders.

Control Signal

DMX512

Developed by the Engineering Commission of United States Institute for Theatre Technology (USITT), the DMX512 standard (For "Digital Multiplex with 512 pieces of information") was created in 1986. It is a standard for digital communication networks that are commonly used to control stage lighting and effects. It was originally intended as a standardized method for controlling light dimmers, however, it soon became the primary method for linking not only controllers and dimmers, but also more advanced fixtures and special effects devices such as fog machines and moving lights. DMX512 does not include automatic error checking and correction, and so is not an appropriate control for hazardous applications. A DMX network employs a multi-drop bus topology with nodes strung together in what is commonly called a daisy chain. A network consists of a single DMX512 controller - which is the controller of the network and one or more client devices.

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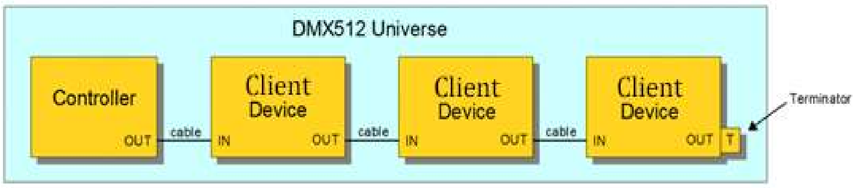
Each DMX512 network is called a "DMX universe" and contains 512 assignable addresses. Each OUT connector on a DMX512 controller can control a single universe. Smaller controllers may have a single OUT connector, enabling them to control only one universe, whereas large control desks have the capacity to control multiple universes, with an OUT connector provided for each universe.



Each client device has a DMX512 "IN" connector and usually an "OUT" (or "THRU") connector as well. The controller, which only has an OUT connector, is connected via a DMX512 cable to the IN connector of the first client. A second cable then links the OUT or THRU connector of the first client to the IN connector of the next client in the chain, and so on. For example, the block diagram below shows a simple network consisting of a controller and three clients.

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Dimmers

Dimmers are devices used to vary the intensity (brightness) of a light. By changing the voltage waveform applied to the lamp, it is possible to vary the intensity of the light output. Although variable-voltage devices are used for various purposes, the term dimmer is generally reserved for those intended to control light output from resistive incandescent, halogen, and more recently compact fluorescent lights (CFLs). So not L.E.D.s!

Dimmers range in size from small units the size of a light switch used for domestic lighting to high power units used in large theatre or architectural lighting installations.

In the professional lighting industry, changes in intensity are called “fades” and can be “fade up” or “fade down”.

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2. BASIC FIXTURE TYPES

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Profiles (AKA ERS, Ellipsoidal Reflector Spotlight, Leko)

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- Profiles get their name from their ability to project the shape of anything placed in the gate of the lantern between the lamp and the lens.
- These shapes may be formed by the **shutters**, or they may be cut out of a thin piece of metal (a "**gobo**"). Gobos are metal cutouts or metal etched onto glass, which are used in a gobo holder to project a defined shape or two break up a beam in a particular pattern. Like a stencil for light.
- You can't use barndoors on a profile as the shutters do a better job, and the way the optics of the profile work, the barndoors would not work anyway.
- A zoom profile lantern is known by the range of its beam angle (e.g. Prelude 16/30, Cantata 18/32 are both zoom profiles from Strand Lighting's range).
- The beam size can be reduced even further by the use of an **iris diaphragm**. This is inserted into the gate of the profile (where the gobo holder would go, so both can't be used together) and features an adjustable aperture which can cut the beam down to almost nothing.

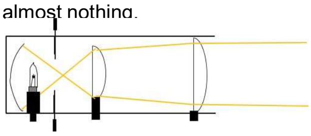

Figure 3 - Diagram showing light beam in a profile (Narrow - lenses far apart)

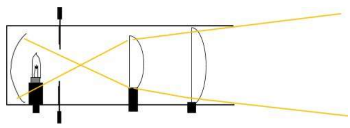

Figure 4 - Diagram showing light beams in a profile (Wide - lenses close together)

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Fresnel (pronounced "Frennel")

- A soft-edged spotlight with more control over beam angle than floods, but less control than profiles.
- The lens is a series of stepped concentric circles on the front and pebbled on the back.
- The size of the beam can be adjusted by moving the lamp and reflector closer to or farther from the lens, either by a screw mechanism or a simple slide.
- The beam can be shaped by the four barndoors attached to the front of the lantern. (NO Shutters, NO Gobos)

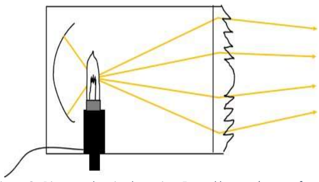

Figure 6 - Diagram showing beam in a Fresnel lantern (narrow focus)

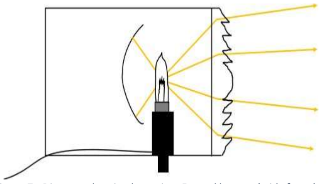

Figure 7 - Diagram showing beam in a Fresnel lantern (wide focus)

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Parcan (AKA Par 64, Par, Can)

- The lantern itself is simply a "can" in which the PAR lamp is contained (hence "Parcan").
- The PAR (Parabolic Aluminised Reflector) lamps are available in a range of beam angles (See pictures)
- The lamp is a sealed beam unit consisting of a lamp, reflector and lens in one.
- The beam is an elliptical shape because of the shape of the filament, and can be rotated simply by rotating the lamp. Access to the lamp is via the rear of the lantern.
- Although they're not widely used, barndoor accessories are available for Parcans. (NO Shutters, NO Gobos).
- Parcans are not suitable to use for general front wash lighting except for music or comedy venues where the amount of light and inconsistent beam are not a problem.
- Many venues are replacing Parcans with **LED** units as they are far more energy efficient, and enable an (almost) infinite variety of colours to be produced.

Figure 8 - CP60, Clear Front

Figure 9 - CP61, Frosted Front

Figure 10 - CP62, Elongated blisters

Figure 11 - CP95, Small squares

Figure 12- LED, NOT a Bulb/Lamp.

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Fixtures common in TV/Broadcast

Red Heads

Red Head is a generic term to describe a range of lights that share two main qualities

1. They can adjust between spot and flood
2. They typically use an 800W tungsten halogen globe

The example pictured here is an Arrilite 800w open-face focusing tungsten floodlight. The beam is focused using the yellow control at the back — this adjusts the reflector rather than the lamp, which should give the lamp a longer life because it is not being moved.

Blonde Lights

Blondes are typically bigger, brighter redheads. Power rating can be 1000 to 2000 watts, although the term generally refers to a 2000w open-face unit.

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Softboxes

A soft box is an enclosure around a bulb comprising reflective side and back walls and a diffusing material at the front of the light. The sides and back of the box are lined with a bright surface - an aluminized fabric surface or an aluminium foil, to act as an efficient reflector. In some commercially available models the diffuser is removable to allow the light to be used alone as a floodlight or with an umbrella reflector.

Advantages of Tungsten Lights

- Almost perfect colour rendition
- Low cost
- Does not use mercury like CFLs (fluorescent) or mercury vapor lights
- Better colour temperature than standard tungsten
- Longer life than a conventional incandescent
- Instant on to full brightness, no warm-up time, and it is dimmable.

Disadvantages of Tungsten Lights

- Extremely hot
- High power requirement
- The lamp is sensitive to oils and cannot be touched
- The bulb is capable of blowing and sending hot glass shards outward. A screen or layer of glass on the outside of the lamp can protect users.

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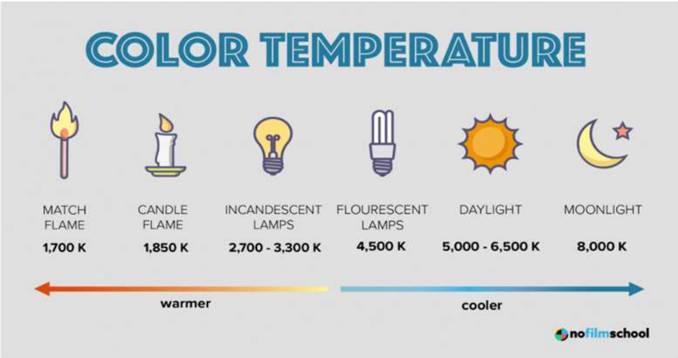
NEW SKILLS 4 NEW ARTISTS

Colour Temperature

Colour temperature describes the "colour" of white light by a light source radiated by a perfect black body at a given temperature measured in degrees Kelvin. Light can be warm (yellow/orange) or cool (blue). Colour temperature can even affect the tone of your story. The best lighting uses both cool and warm to create an environment and to get the colour you want out of an object or subject.



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3. BASIC LIGHTING ANGLES

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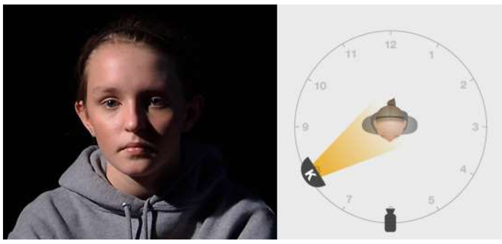
Basic Lighting Angles

Three-point lighting is a traditional method for illuminating a subject in a scene with light sources from three distinct positions. The three types of lights are key light, fill light, and backlight.

- **Key light** - This is the primary and brightest light source in the three-point lighting setup. It gives a scene its overall exposure. Cinematographers typically position this main light slightly off to the side of the camera and the front of the subject, on a light stand at a 45-degree angle to the camera, which creates shadows on the opposite side of the subject's face, giving it dimension and depth. The primary light creates the mood of a scene. Depending upon its position and the supplemental lights used in the overall lighting, it can create a high-key image (evenly, softly lit and atmospherically upbeat) or a low-key image (high contrasts, deep shadows, and very moody).



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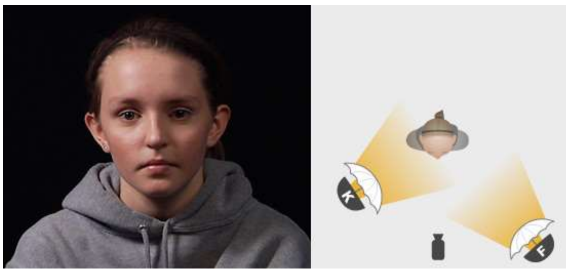

- **Fill light** - Mirroring the key light on the opposite side of the camera, the fill light literally fills in the shadows that the key light creates on a subject, bringing out details in the darkness. Typically, this secondary light is less bright than the key, and cinematographers control the overall feel of their shots based on how much they dim or lighten the fill light. A dim fill light, where the fill ration is high, creates a high-contrast, film-noir type of shadow, while a brighter light with a lower, more balanced ratio gives the subject a more even look. The second light isn't always a light: it can be a reflector, a bounce card, a wall, or anything that bounces back some light onto the subject to fill in the shadows. Together with the key light, the fill light determines the mood of a scene.

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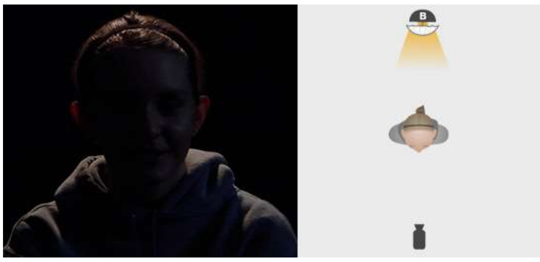
- **Backlight:** The third source in this lighting technique, the backlight (also known as the “rim light” or “hair light”) shines on a subject from behind, completing the light setup. This creates a rim of light or outline around their head that pushes the subject away from the background and gives a sense of depth. Typically, cinematographers position the backlight directly behind the subject or high enough to be out of frame, opposite the key light, and pointing at the back of the subject’s neck.



There is no set formula for how three-point lighting is used. This often depends on the scene, the subject matter, and the overall mood that a cinematographer or photographer wants to evoke.

- Good lighting creates a more interesting and dynamic image where the subject is seen with more dimension and where the cinematographer has more control over shadows.
- By placing a soft key light slightly off center with a 2:1 fill ratio, a cinematographer creates a soft, flattering look that also tends to hide blemishes in the skin when your subjects are people. This soft lighting is called “high key lighting” and creates an optimistic, upbeat, youthful, light, and airy mood that is common in sitcoms and comedies.

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Tips for Setting Up Three-Point Lighting

- Establish your light’s “motivation.” Before you begin setting up your lighting kit, you have to know precisely what look you want to achieve and why. Lighting setups are never random. The source of light in your scene needs to make sense based on the environment that your characters exist in. Is it an overcast sky? Sunset? A dark alley? Once you have established the motivation, you can proceed to place and adjust your light kit to achieve that effect.

- Consider light source size and distance. The size of a light source relative to the subject size determines how “hard” (sharp, distinctive edges) or “soft” (smooth, feathered edges) your shadows will be. A smaller light source creates harder, distinct edges, while a bigger one softens the shadows. In studio lighting, if you want a softer look, you place enlarging modifiers such as an umbrella, softbox, or another diffusion between the light source and the subject. Due to this relative size condition, the distance of the light source to the subject will also affect shadow softness

- Consider the intensity of your light source. “Brightness” is the measure of a light source’s intensity. You measure it in lumens with a light meter. With LED lights, fluorescent lights, and incandescent lights, you control the output intensity, which affects the look of your scene. Brighter light will create harsher edges and shadows.

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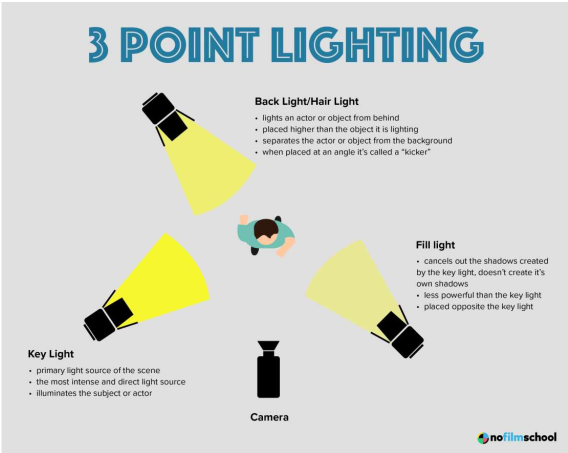
- Consider the position of your light sources. Where you place your lights relative to your subject and the camera determines where shadows fall. This relates again to sensibly creating an environment—if your key light represents the sun, it should accurately reflect the angle and height of that source. How you position your fill and backlight affects whether there are deep, moody shadows or an optimistic, even light cast across your scene.

- Test your setup. After you have determined your lights' motivation, their size, distance, intensity, and position, set everything up so you can see exactly how all the lights work together and whether or not their effect is precisely what you intended it to be. If it is not, make adjustments until everything is perfect.



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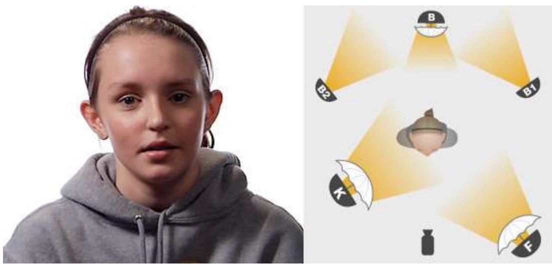
Lighting Backgrounds

- Now that our subject is properly lit, we can experiment with the backdrop. Let's try to achieve a popular backdrop, which is a solid bright-white backdrop, free of shadows and detail.
- A white sheet has been hung on the wall about 6 feet behind our subject, but we can obviously see that it's a sheet (and in need of a good iron).
- Why doesn't the sheet look white? The reason is that the camera's exposure has been set according to the subject, who is much more brightly lit than the backdrop. Simply put: we need more light on the backdrop.



Two lights will give good even coverage- one on each side. It's important to note that the lights have been positioned in such a way so that their light does not spill onto our subject. Remember - she has been properly lit, and we don't want extra light ruining our setup. There are devices called barn doors which can be put on lights to prevent extra light from spilling in undesired directions. In this case, it was sufficient to just put the lights low on the floor behind our subject and point them at the wall.

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4. DRAWING A PLAN

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NEW SKILLS 4 NEW ARTISTS

The Basic Lighting Plan

As with any project you undertake you will need some kind of formulated approach, in the case of lighting design that can be presented in a number of documents, one of the most important of these being the lighting plan. Lighting plans can range in complexity from a massively intricate A0 printed plan or CAD (Computer Aided Design) document, to something as simple as a quick sketch on the back of a beer mat (I've literally had this handed to me). The most important part of the plan is to get across the concept of the design and the key information that the technicians or other people working on the project can understand your intention as the designer.

We have already looked at the basic 3 point source lighting methods that you may use for a small to medium size streaming setup, now we will see how this knowledge will be scaled up to be used in a larger size production or broad cast and how we would go about documenting the process.

Before we begin there are some things to bare in mind:

- What is the location of the performance?
- What kind of performance is it?
- What equipment will I have at my disposal?
- What facilities are there to hang or rig lights?
- Is there an adequate power supply?

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1. A scale plan of the set or venue: This almost always includes architectural features of the stage as well: rear and side walls if they are close enough to be relevant, and the sides of the proscenium arch. However if you are given a plan of the set which does not include these, you will have to add them, to the same scale. The idea is to have a scale plan which includes all important physical features of the area you need to light.
2. A ceiling plan of the theater:
   - If the theater has a fixed grid this should be a plan of the grid.
   - If it has electric pipes which are raised and lowered you will need to know where all the pipes are.
   - The ceiling plan must be to scale; if it isn't, you will need to find out the exact locations of the lighting positions so that you will be able to add them to scale in your lighting plan.
   - You will also need to know the exact height of all lighting positions. If the theater has a fixed front of house bridge this information might be difficult to obtain, but you should try. In a properly designed theater the angle for front of house lighting is about 45 degrees but there are theaters which are not well designed and the angle might be steeper or shallower. If you can't get exact information, you might be able to turn on a fixture from the front of house bridge, and then you will be able to measure the size of area it covers.
3. A list of lighting elements: You will have prepared this during and probably revised it and cleaned it up, unifying some elements, adding others, following talks with the director and designer and further thought. This will be within the spec of equipment available in the venue unless you have a budget to rent additional equipment.

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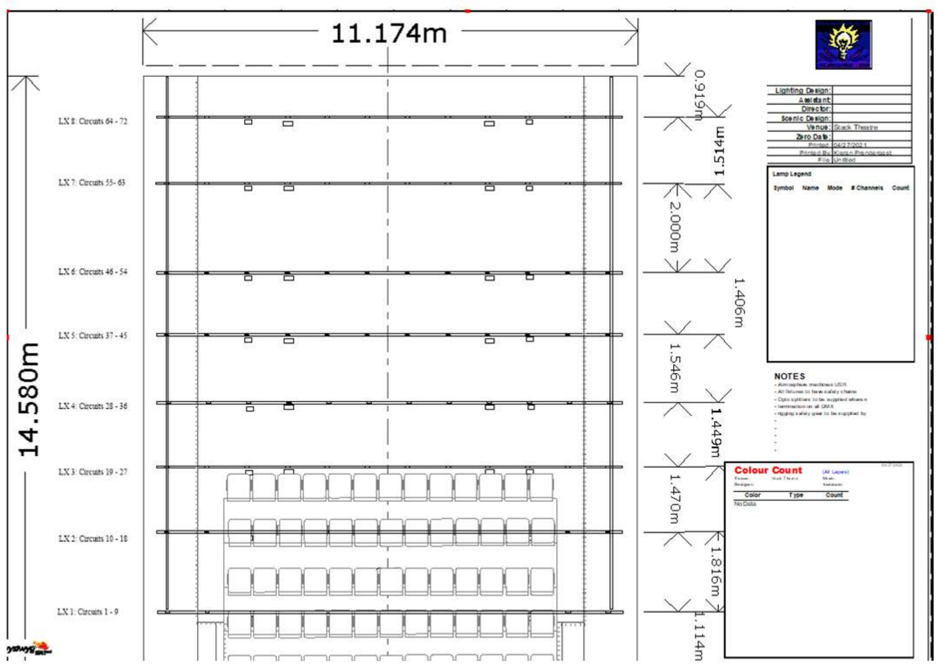

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Simple 3 Point Source Plan

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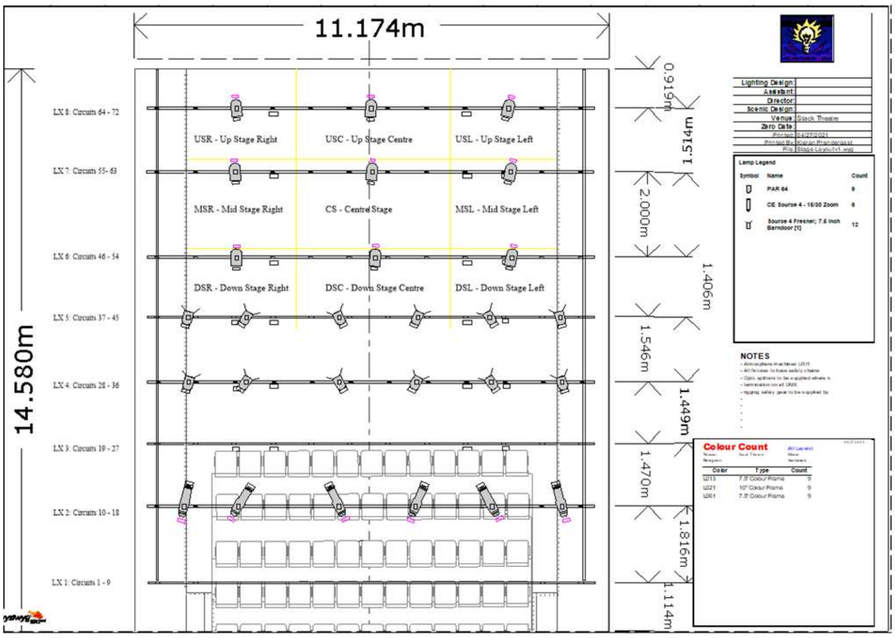

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NEW SKILLS
4 NEW ARTISTS

A Basic Gig layout:

1. Put your band in position.
2. Light the band from the front (most ticket buyers like to be able to see the performers), 3 point source is not always necessary for live music events so a fill light may not be needed.
3. Add you're your backlights, generally with very saturated colours – these will provide colour washes of the stage and set the general mood for each number.
4. If you have the option for a follow spot on lead vocal, make use of it, it means the operator can concentrate on making the show look good.
5. Add effects lighting, this can be a combination of generic fixtures, automated fixtures or whatever else is available, what you use will also depend on the type of event it is, so perhaps no strobe for a classical recital?

Things to remember:

If you are lighting orchestras or classical performers it is likely they might want to see a score so part of your job is to facilitate that. This can be done using a dedicated lighting fixture or with a music stand light depending on the type of event or the performers choice.

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NEW SKILLS 4 NEW ARTISTS

5. PRACTICAL EXERCISE

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NEW SKILLS 4 NEW ARTISTS

Learning Outcomes:

1. Be able to identify different fixtures.
2. Draw a basic plan for a simple 3 point source setup for a single performer.
3. Setup a simple 3 point source lighting setup for a performer.
4. Identify the different positions by name.
5. Setup a camera to see the difference between what you see and what is displayed on a visual monitor.
6. How does altering the proximity of the fixtures affect what you see?

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6. ADDITIONAL CONTENT

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Introduction to Automated Lighting

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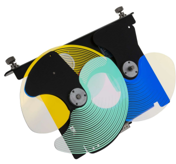
4 NEW ARTISTS

Automated lighting is very popular within the industry because of how versatile it is. Instead of having to rig and focus many generic (non-automated) fittings with different coloured gels or gobos, designers can now use one light to get the colours they desire, with the benefit of having many more features. (Such as gobos, the ability to move the lights for visual effect.) Combining the differing properties in different ways gives the user the ability to create different looks, feel and effects using just one fitting.

Originally, automated lights were commonly known as "intelligent" lights. Many in the lighting industry felt this was mis-leading as this type of fitting does not do any "thinking", they do what they are "told" to by the control desk, therefore the term “Automated” was used. This is done via the DMX512 protocol. Almost all automated lights have many and various functions and capabilities. For example, an automated light might have several different colours (on a colour Wheel or Colour Mixing) , pan and tilt functions, zoom, gobos, and shutter options. Usually each of these properties is assigned a DMX512 address by the firmware in the light itself.



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NEW SKILLS 4 NEW ARTISTS

Automated fixtures usually employ compact arc lamps as light sources (newer ones use LEDs) and require a constant 240v supply of power. They use servo motors or, more commonly, stepper motors connected to mechanical and optical internal devices to manipulate the light before it emerges from the fixture's front lens. Examples of such internal devices are:

- Mechanical dimming shutters used to vary the intensity of the light output as the lamp does not dim electrically. Mechanical dimmers are usually a specially designed disk or a mechanical shutter. Shutters with high speed stepper motors can be used to create strobe effects.
- Pattern wheels with gobos and *gate shutters* to change the shape of the beam or project images. Some fixtures have motors to rotate the gobo in its housing to create spinning effects, or use their complicated lens systems to achieve the same effect.

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Fixture Profile Library

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A DMX profile or Fixture profile is set up so that your console is able to communicate with your lights. This communication will tell your console exactly what type of fixture you have and what DMX commands the fixture is capable of. DMX is really a one way street when it comes to your console and your fixtures communicating with each other. The console just wants to read what type of light you have and what functions it can do? There's really no communication that goes from the light back to the console, although some newer protocols like DMX RDM allow some receipt of information from the fixture.

This is why it's so important to be able to set up your DMX fixture profile correctly and allow it to communicate with your console. Most consoles have a vast library of profiles built in that gets updated with every subsequent software revision but sometimes you may have to build one yourself.

Building a DMX Profile

The first step to setting up or building a Fixture profile for your fixtures is getting the manual for your lights. This should tell you exactly how to set up the DMX fixture profile.

On the right we see a list of parameters for an LED fixture which requires 9 channels of control to function as it has 9 parameters/attributes.
For a lighting desk to know what parameters these are and what order they are in we must tell it by creating a Fixture profile.
If we mix up any of the parameters it could effect the fixture in a number of different ways ie. If we put the colour channels in the wrong order the colour picker function on the desk won't assign the correct colour to the fixture.

9 channel mode		
Channel	**Value**	**Function**
1	000-255	Red (0-100%)
2	000-255	Green (0-100%)
3	000-255	Blue (0-100%)
4	000-255	White (0-100%)
5	000-255	Amber (0-100%)
6	000-255	UV (0-100%)
7	000-255	Master Dimmer (0-100%)
8	000-000	ON
	001-005	Sound control
	006-010	ON
	011-255	Strobe (Speed increasing)
9	000-007	BLACKOUT
	008-255	Colour Macro (30 colours)

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Once you create the profile in the ETC Eos software it will look something like this.

Type	DMX Footprint	Cell
Dimmer	1	
LightmaxX RGBWA+UV Par	9	

LightmaxX RGBWA+UV Par

DMX Footprint: 9 Parameter Count: 12

Create Multicell

Remote Dimmer

#	Parameter		DMX	Home	Snap	Ranges
1	Red	Color	8b: 1	255		0 > 100
2	Green	Color	8b: 2	255		0 > 100
3	Blue	Color	8b: 3	255		0 > 100
4	White	Color	8b: 4	255		0 > 100
5	Amber	Color	8b: 5	255		0 > 100
6	UV	Color	8b: 6	255		0 > 100
7	Intens	Intensity	8b: 7	0		0 > 100
8	Shutter Strobe	Form	8b: 8	0	✔	4 Ranges
9	Color Macros	Color	8b: 9	0	✔	0 > 100
10	Brightness	Color	Virtual	255		0 > 100
11	Hue	Color	Virtual	0		0 > 360
12	Saturation	Color	Virtual	0		0 > 100

Once I leave the fixture builder, patch the fixture, and return to live mode I can then access the parameters of the fixture for programming. Below you can see the first of 3 pages of colour settings I can access once I select the fixture Channel number.

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The number and variety of different manufacturers, models and types of moving light is staggering. Depending on the production's budget, and the designer's preferences, the moving lights chosen for a production's lighting rig are almost limitless.

While the inclusion of moving lights into the design opens up a world of possibilities in the design, it also increases exponentially the complexity of the design, and so the inclusion of moving lights during the design process should be carefully considered with all of the other limitations of time and budget. Here are some examples of different types available.

The moving head fixture is the most common moving light available in today's market. Movement includes the ability to actually move the beam of light around the stage, as well as to do things like spinning gobos, scrolling color wheels and other effects. Every moving light has its own special features: some have controllable shutters, others are able to zoom larger and smaller, some have prism effects. Some instruments only have a limited number of colors attached to a fixed color wheel, and others have CMY color mixing through which the designer has almost limitless possibilities in color choice.

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- Another way of moving light around the stage is by using a moving mirror. In this type of unit, the instrument itself does not move, but a mirror placed in the path of the beam is attached to motors and bends the beam of light to its intended target. Moving Mirror fixtures had been quite popular during the advent of the popularity of moving lights. However, moving head fixtures proved to be more popular, and moving mirrors are not used or manufactured much anymore. You might find them in some theatres, as they were less expensive than the moving head units in the 1990s, so they are worth knowing about.

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Pros

There are many pros to using automated lighting:

- Time saving:
  - Focus can be done from the desk.
  - Fewer fitting required to rig as gobo washes or specials.
  - Fewer power cables to run.
- Aesthetics:
  - Far more looks available to the operator or designer, this is the major reason for using them.

Cons

There are also a few cons to consider:

- Expense: automated lighting costs far more than generics to purchase or hire. An experienced technician is required to program automated lights.
- Weight: usually far heavier than an average generic.
- Reliance on a DMX chain: one malfunctioning DMX lead can prevent an entire rig from functioning.
- High maintenance: automated lights have many moving parts and so require a good maintenance schedule.

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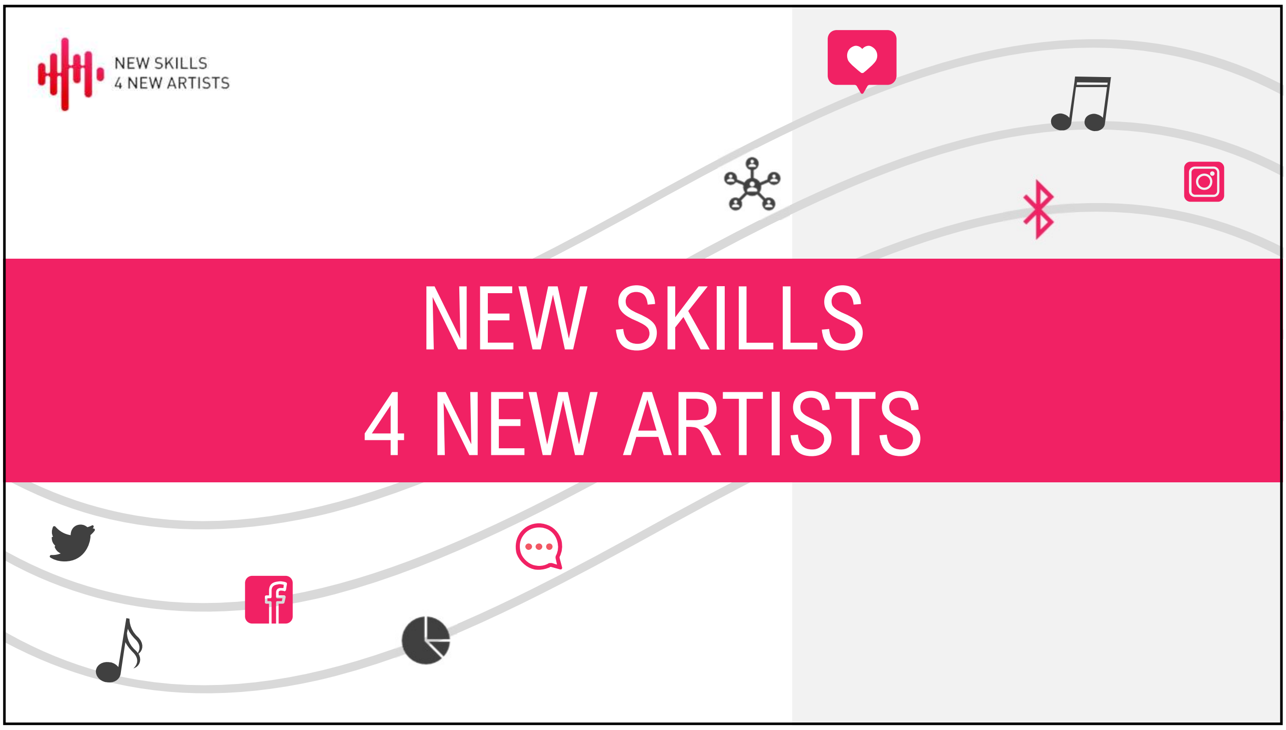
NEW SKILLS
4 NEW ARTISTS
NEW SKILLS
4 NEW ARTISTS

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MODULE 2
NEW SKILLS
4 NEW ARTISTS
INTRODUCTION
TO
TECHNOLOGY
(AUDIO VISUAL)

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UNITS

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1. CAMERAS

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Which cameras are suitable for live streaming?

- How do you know a camera is good for live streaming?
- Can you use the one you already own?
- How do I connect my camera?
- What are the optimal settings for my camera?
- What kind of Lens does my camera need?
- Can I use an iPhone/Android camera to stream?
- How do I set a white balance on my camera?

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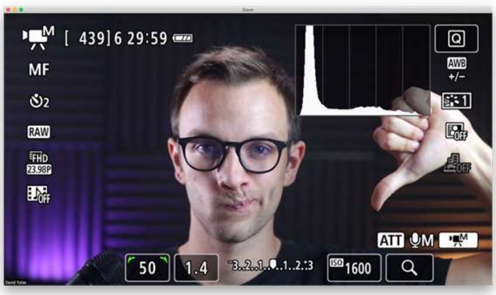
Four essential criteria for live streaming cameras

1. Clean HDMI out

To live stream from a camera, you have to capture the signal coming directly from its HDMI or SDI out port. Along with the video feed, some cameras will also send all the user interface (UI) elements visible on the display (e.g., battery life, exposure, aperture). To be suitable for live streaming, your camera has to be capable of sending a "clean" signal over HDMI, i.e., a signal without any UI elements visible. Unless it's clean by default, there should be a menu setting you can toggle.



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2. Power supply / AC adapter-ready

Live streams can run for hours. Most internal batteries can only last for about 20 minutes. Make sure there's an option to get an AC power adapter for your camera (and get it!).

3. Unlimited runtime

For safety and battery conservation reasons, some cameras (especially DSLR models) will automatically shut off after about 30 minutes of inactivity. Automatic shutoff will not be acceptable for longer live streams. Check to see if your camera has this safety feature and whether there's a way to disable it in settings.

4. No overheating

If you are planning to stream for over an hour, camera overheating may become an issue. Some mirrorless and DSLR cameras can overheat, especially when powered over USB. One way to prevent this is to use something called a dummy battery and an AC power adapter instead of USB power. Even so, some cameras are just more prone to overheating than others. Be sure to research this before buying.

Be it a DSLR, a camcorder, cinema, mirrorless, or any other type, if your camera meets these four criteria, your camera is ready for live streaming. Webcams, on the other hand, are designed specifically for streaming, so it's safe to assume that most of them come out of the box ready to live stream. It is also safe to assume that all camera models listed in this article comply with these guidelines.

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Logitech Brio 4k

Canon M200 mirrorless

Canon EOS 90D DSLR

Sony Handycam Exmor FDR-AX53

Blackmagic Pocket Cinema Camera 6K Pro

RED Digital Cinema DSMC2 Dragon-X

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NEW SKILLS 4 NEW ARTISTS

2. CONNECTING A CAMERA TO A COMPUTER

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NEW SKILLS 4 NEW ARTISTS

Connecting a Camera

Producing high-quality live streams is one of the best ways to gather millions of followers in a short period of time, but in order to achieve this goal, you must have the adequate equipment.

Do I Need A Capture Card for Streaming?

Whether or not you'll need a capture card to start streaming depends on several factors. Desktop and laptop owners can easily start a live stream with a screencasting software that is usually more affordable than a capture card (Or using software like OBS – Open Broadcaster Software. More on this later!). Still, you may encounter some limitations if your PC's configuration is not powerful enough to support live streaming at large video resolutions.

Capture cards let you have complete control over the stream's quality, but broadcasting live video in 4K resolution is not possible on Twitch, because the platform still doesn't offer this option, however it is possible for some users on YouTube. Many users still only choose to stream in Full HD (1920x1080p) as the bandwidth and processing requirements are much lower.

When purchasing a capture card pay attention to the resolution at which the card can capture the video.

- HD Ready – 1280x720p @60 fps
- Full HD – 1920x1080p @60 fps
- Ultra HD – 3480x2160 @60 fps

The higher the resolution the better the stream quality, this also means more processing power is necessary in your computer and higher bandwidth to upload your stream.

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NEW SKILLS
4 NEW ARTISTS

HDMI Capture Devices

Elgato Cam Link 4k HDMI Camera Conn

- For connecting a DSLR, a camcorder or an action cam to a PC or Mac
- Supported resolutions: 3840x2160 up to 30p, 1920x1080 @60p/i, 1280x720 @60p/i

Generic Video Capture Card

- For connecting a DSLR, a camcorder or an action cam, PS4, Xbox to a PC or Mac
- Supported resolutions: 3840x2160 up to 30p, 1920x1080 @60p/i, 1280x720 @60.p/I
- Plug & play capture card, no driver needed, so you can use the built-in functions of the current software to get a seamless experience. Especially suitable for beginner.

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NEW SKILLS
4 NEW ARTISTS

SDI Capture Devices

- Blackmagic Design DeckLink Duo 2;
- PCI Express capture and playback card; with four separate 3G-SDI connections; supports formats in SD and HD up to 1080p60.
- Operating systems: Mac Mojave 10.14, or higher; Windows 8.1 and 10, both 64 bit; Linux.

Blackmagic Design UltraStudio 4K Mini

- With Thunderbolt 3, 12G-SDI, HDMI 2.0 and analog connections for broadcast-compatible 8, 10 and 12-bit recordings of high dynamic range videos in all formats from SD to 4K DCI at 60 frames per second.
- Analogue audio inputs: 2 channels of professional balanced analogue audio via 1/4" jack connections
- Analogue audio outputs: 2 channels of professional balanced analog audio via 1/4" jack connections
- SDI audio inputs: 16 channels embedded in SD / HD / 2K / UHD / 4K
- SDI audio outputs: 16 channels embedded in SD / HD / 2K / UHD / 4K

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3. OPTIMAL CAMERA SETTINGS

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Optimal Camera Settings

To get the best possible video quality, be sure to check the following settings before going live. Some of these settings will be easily adjustable using dials on the camera body, while others may be hidden deep within the system menu or within the computer software.

Coordinate exposure and fps

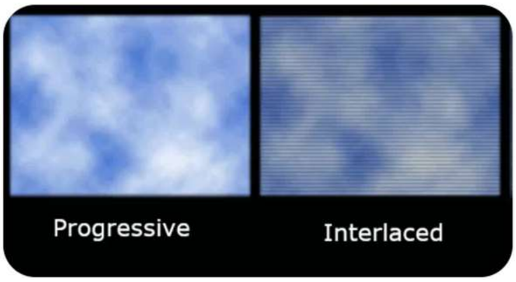
Set the exposure time fraction to be a multiple of the chosen frame rate. For example, if you are streaming at 30 frames per second (fps), set your camera's exposure to be 1/30; at 60 fps make it 1/60; etc.

Fully open aperture

The more light the camera receives, the better the image quality, as more light is able to hit the sensor.

Progressive, not interlaced

Video sources that are listed with the letter "p" are called progressive scan signals (e.g., 720p, 1080p), while those listed with "i" are interlaced (480i, 1080i). Progressive signals look better because they display both even and odd scan lines simultaneously. Interlaced signals alternate between even and odd scan lines, making the video look stripey. Always go for "p" for streaming, not "i".



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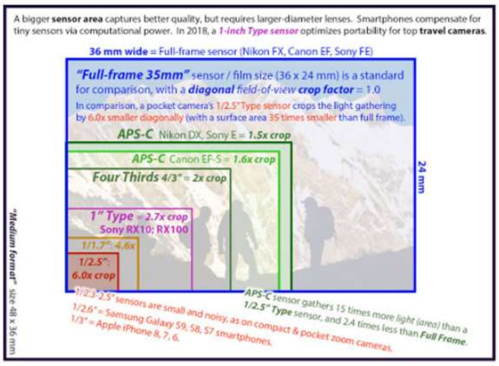
How to Choose a lens

When using a DSLR (digital single-lens reflex camera) or Professional grade camera you will usually need to do a little research as to what lens is going to suit your requirements. To do this you will need to familiarise yourself with the type of system that connects the lens to the camera and also the sensor size of your camera. Common mounts you will come across are:

- Canon EF- and EF-S-mounts – For 35mm, Full Frame or APS-C size sensor.
- Micro Four Thirds mount – For a Micro Four Thirds Sensor.
- Nikon F-mount - For 35mm
- Pentax K-mount – APS-C size sensor



Aside from the mount and sensor you will also want to consider the distance from the subject to the lens, the amount of light that the lens can capture and the focal length of the lens. Here are some links that will give you further information on all of these things:

https://www.learnaboutfilm.com/making-a-film/equipment-for-low-budget-filmmaking/choosing-lenses/

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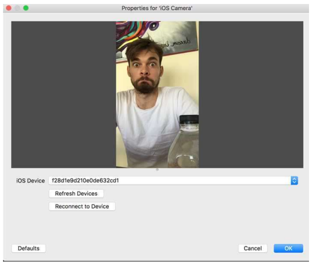
How To Live Stream Using Your iPhone

There are a many apps that enable live streaming for mobile devices. There are also a few that are compatible with several different live streaming platforms, but while streaming directly from your phone to your platform of choice might be convenient, it doesn't produce the polished, slick productions that we have become accustomed to seeing online. That doesn't mean we can't still make use of the camera in our phone, there are a number of apps that can allow us to connect our phone camera to our computer.

We will be looking at OBS as our broadcast software and they have an IOS and Android app that can be used with your phone once it is plugged into your computer. For this to work you will also need to install a plugin on your computer to allow you to add it to OBS.

When you connect your phone make sure it is orientated in landscape mode or your stream will look like this. >>>



Here is a link to help you get setup if you wish to use your phone as your camera:

https://obs.camera/docs/getting-started/ios-camera-plugin-usb/

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Setting a White Balance

NEW SKILLS 4 NEW ARTISTS

What Is White Balance?

- In non-technical terms, white balance is how warm or cool the overall colors in your photograph look.
- Usually, colors in your photos will look pretty close to the way they looked in real life. However, your camera is easily confused and can sometimes make the colors too warm or too cool.
- The most obvious place to spot this problem is the parts of your scene that are, or should be, white. When you take a photo by candlelight, sometimes the whites will look kind of yellow or orange. On a cloudy day, or when you're in heavy shade, the whites might look a little blue. Sunlight in the morning and evening can make colors a little redder or "warmer".



- This is called a "color cast," and it happens because the color of the light source varies. Ambient light on an overcast day can be a little bluer or "cooler". This warmth or coolness in the colors is referred to as "color temperature".

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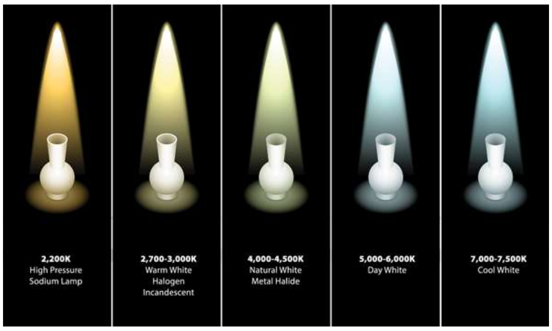
Understand Color Temperature

NEW SKILLS 4 NEW ARTISTS

Color temperature is measured in degrees Kelvin. Different sources of light have different color temperatures. Incandescent or tungsten lights are warm. Candlelight is even warmer. The natural light on a cloudy day is cooler, while fluorescent light can give your photo a green cast.

Here's a brief rundown of some of the most common lighting situations you might encounter and what the corresponding Kelvin number is:

- Candlelight: 1900
- Incandescent light: 2700
- Sunrise/golden hour: 2800 to 3000
- Halogen lamps: 3000
- Moonlight: 4100
- White LEDs: 4500
- Mid-day: 5000 to 5500
- Flash: 5500
- Overcast/cloudy: 6500 to 7500
- Shade: 8000
- Heavy cloud cover: 9000 to 10000
- Blue sky: 10000



Note that the cooler the light, the higher the number. The warmer the light, the lower the number.

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NEW SKILLS
4 NEW ARTISTS

Use Auto White Balance

Most cameras default to the “Auto” white balance setting, which actually works pretty well, most of the time. In auto white balance mode, your camera examines the scene you’re trying to photograph and chooses a color temperature (in Kelvin) it thinks will work best.

However, your camera can easily get confused if the scene:

- Doesn’t contain any colors which are white, or close to white.
- Contains mostly one color (e.g. a lot of green grass, white snow, blue sea or sky.)
- Is illuminated by multiple light sources with different color temperatures.

All of these scenarios can result in a color cast in your photo, and you’ll want to take charge of the white balance.

Set Your White Balance Manually

For tricky lighting situations, including “mixed lighting”, you’re going to get the best colors if you ignore the presets or auto altogether and set your white balance manually. You’ll still end up with a single color temperature being applied to the entire scene, but the results will be better than can be achieved with the presets.

In general terms, setting white balance manually involves taking a photo of something white or mid-gray in the same light which is illuminating your intended subject. Next, you select your camera’s Custom White Balance mode, and tell the camera to use the photo you just took of the white or mid-gray content as a reference.

Here is a short video on White Balance: https://www.youtube.com/watch?v=g1lD6fT_DEo

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NEW SKILLS
4 NEW ARTISTS

4. CREATING A LUT

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NEW SKILLS
4 NEW ARTISTS

Creating a LUT in Da Vinci Resolve

- You can download Da Vinci Resolve here (The base version is free): https://www.blackmagicdesign.com/products/davinciresolve/
- Before we start you will need a picture from your webcam in the environment in which you will be streaming with the lighting you will use.
- You will also need an Original.png LUT from here: C:\Program Files\obs-studio\data\obs-plugins\obs-filters\LUTs.

1. Open up Resolve and start a new project.
2. Select the second tab on the bottom of the screen (Cut), then drag both the Original LUT file and the Webcam still that you captured into the Media Pool.
3. Next we will move to the fifth tab at the bottom of the screen, Colour, we will be able to do any necessary colour grading within this tab (there are many videos available on this subject online).
4. You will use the colour controls on the bottom left of the screen to adjust the colour of the image.
5. Once you have finished making adjustments go to the cog/gear wheel on the bottom right of the screen to open your project settings.
6. Open the “Master Settings” where you will see “Timeline Resolution”, change these to 512x512. Save your project settings and close the window.
7. The image displayed changes size, we will now select the Original.png file so it will be displayed, from here we will middle click on the still we captured from our camera. When we do this the colours will change, this means we did it correctly.

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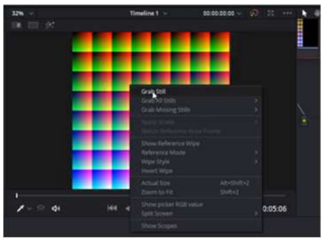
NEW SKILLS
4 NEW ARTISTS

8. We will then right click on the Original.png file from the main video window and select “Grab Still” (see image).
9. The grabbed still will appear on the top left of the screen in the stills browser window.
10. Right click on the still and select “Export”, select a location (C:\Program Files\obs-studio\data\obs-plugins\obs-filters\LUTs), and set file type to .png.
11. Click “Save”.
12. You have now created a LUT file for your specific camera.



LUTs can be used with most cameras and are an easy way of saving colour settings to recall them in a variety of software at a later point.

A guide for creating a LUT using Da Vinci Resolve: https://www.youtube.com/watch?v=tTj-uvVrOYM

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NEW SKILLS 4 NEW ARTISTS

5. INTRODUCTION TO OBS

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NEW SKILLS 4 NEW ARTISTS

Getting Started with OBS

OBS is a free and open source solution for offline video recording and live streaming . It is available for Windows, macOS, Linux. With an open canvas approach to video creation this tool can mix a variety of audio and video sources to a single output for creative video and broadcast applications. OBS has the ability to record straight to your hard drive using a record only mode or in conjunction with a simultaneous live stream.

- You can download OBS Studio from here: https://obsproject.com/download

What we will look at in this session:

- Opening OBS and getting started.
- The interface layout (All the different parts, and where to find things).
- Scene Collections – What they are, how to use them, and how to import or export them.
- Scenes – Made up of sources, eg camera, images, audio.
- Sources – Your inputs, camera, images, media sources, capture devices, display capture.
- Scene Transitions – How we switch from one scene to another, cuts, fades, moves, or stingers.

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NEW SKILLS
4 NEW ARTISTS

The Settings Menu

1. Profiles are for setting up Broadcast settings.
2. Scene collections are for having different setups for scenes while streaming.
3. General – No changes
4. Stream – Youtube, add your stream key/ Twitch, log into your account
5. Output –
   a. Encoder settings – h264 unless you have capable GPU
   b. Check upload settings for bitrate.
   c. Keyframe – Important for youtube, set to 2
   d. Recording – set file location for recording, file type to mkv (in case it stops recording mid stream, you will not loose full recording.
   e. Audio – bitrate can be limited on some platforms, I set to 320 and it can get compressed after.
6. Audio – Sample rate – unchanged, choose your devices, Monitoring device is important for zoom.
7. Video – Set your bas canvas size: this can be based on the resolution of your camera, downscale filter – unchanged, frame rate – 30 for youtube.
8. Hotkeys – You can set these up as you wish.
9. Advanced – Mostly unchanged, unless you want to bind your IP address.

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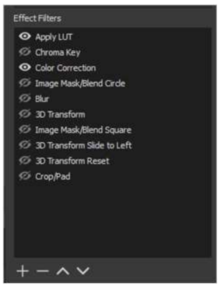
NEW SKILLS
4 NEW ARTISTS

Filters

You can add them by right-clicking your desired Scene, Source or Device and selecting "Filters" (for Audio Devices, click on the gear icon next to your device).

- You can add audio and video filters for each source. Ie Chroma Key for green screen.
- Compressors, gates, eq can be added for your audio sources.



Scene Transitions

- A transition is a technique used in the production process of live broadcast and video editing by which scenes or shots are combined. Most commonly this is through a normal cut to the next shot. Other transitions may include dissolves, cuts, fades (usually to black), match cuts, and wipes.
- Stingers are a type of animated video transition which combine a transparent video animation into a full-screen overlay using a timed cut transition.

For further reference please visit: https://www.youtube.com/watch?v=r7teWxV5BCE

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NEW SKILLS
4 NEW ARTISTS

6. CREATING A LOWER THIRD WITH GIMP

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NEW SKILLS
4 NEW ARTISTS

Using GIMP

GIMP is a free and open-source raster graphics editor used for image manipulation and image editing, free-form drawing, transcoding between different image file formats, and more specialized tasks. In this lesson we will be using GIMP to create a simple lower third for a stream. A lower third is a graphic overlay placed in the title-safe lower area of the screen, though not necessarily the entire lower third of it, as the name suggests. In its simplest form, a lower third can just be text overlaying the video.

You can download GIMP here: https://www.gimp.org/downloads/

- Create new layer for lower third
- Rectangle Select – Rounded corners radius 100 px
- Click hold paint bucket tool to get gradient tool.
- Click and drag for your gradient
- Select text tool
- Select Colour – font – font size
- To add drop shadow – Filters – Light & Shadow – Dropshadow – settings to taste
- Create another lower third layer
- Drag and drop social logos
- To Resize – Right click on layer – Scale layer 65px
- Add social name with text.

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Once you have finished creating your lower third you can use the “Export” function in GIMP to export the lower third as a .png file. Before you export make sure that your base layer is set to be transparent so it sits on top of your video and does not block it from view.

For further reference please see: https://www.youtube.com/watch?v=8jjNrtPdpd4

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