THE HOME RECORDING STUDIO

MANUAL

The Ultimate Beginners Guide To Home Studio Development & Recording

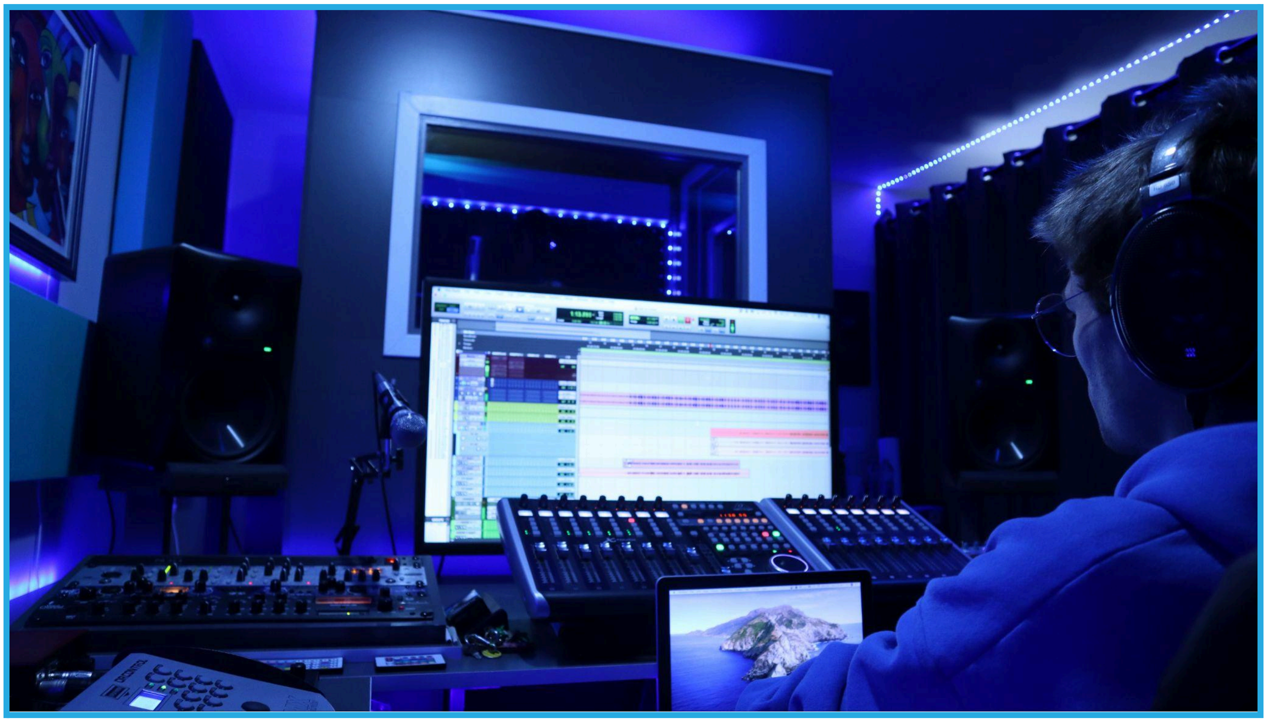
EVERYTHING YOU NEED TO KNOW!

InSession MUSIC TECH

BY JA MIXDOWNS

MODULE 2

Recording Basics Explained



In this module, we will be covering all of the basic fundamentals of the recording process, **which we have broken down into 10 sections:**

1. Understanding the steps of the recording process
2. Signal flow and gain staging
3. Cables
4. Digital audio recording basics
5. Recording software
6. Recording vocals
7. Recording instruments

8. Recording midi
9. Solo Recording

1. Understanding the steps of the recording process

If you wanted to record music in 1996, the chances of you setting up your own home studio and learning to do it yourself were slim to none. Even for the biggest names in music.

Most professional music released to the world was done out of expensive commercial studio's with professionally trained recording engineers and producers that only the big record labels could afford. So even if somebody really wanted to learn how to record and make their own music on a professional level, there was very little access to affordable recording equipment to the everyday average Joe. So the Idea of a home recording studio just wasn't a reality for many.

Fast forward to 2023 and the entire industry of recording has been completely revolutionized and has exploded into a massive billion dollar consumer driven industry, all because of the advancement of technology.

What does that mean?

Anybody can set up a recording studio in the comfort of their own home for very little cost and achieve professional results. We are truly living in incredible times as far as music creation is concerned. Today, more music is being recorded and distributed all around the world than ever before. Musical self-expression is at an all time high!

In theory, because of the Internet and technological advancements it's never been easier to create and have your music heard all around the world. That's the good news. The not so good news is that there's a lot to learn when it comes to home studio development and the recording process. Meaning those professional results aren't going to come overnight. There are probably hundreds of little steps to learn before you can consistently churn out great records but we will break them down into 3 major categories to ease any possibility of being too overwhelmed.

So whether your goal is to become a famous superstar or if you just want to make music for you and some close friends you will still need to understand these 3 steps of the recording process.

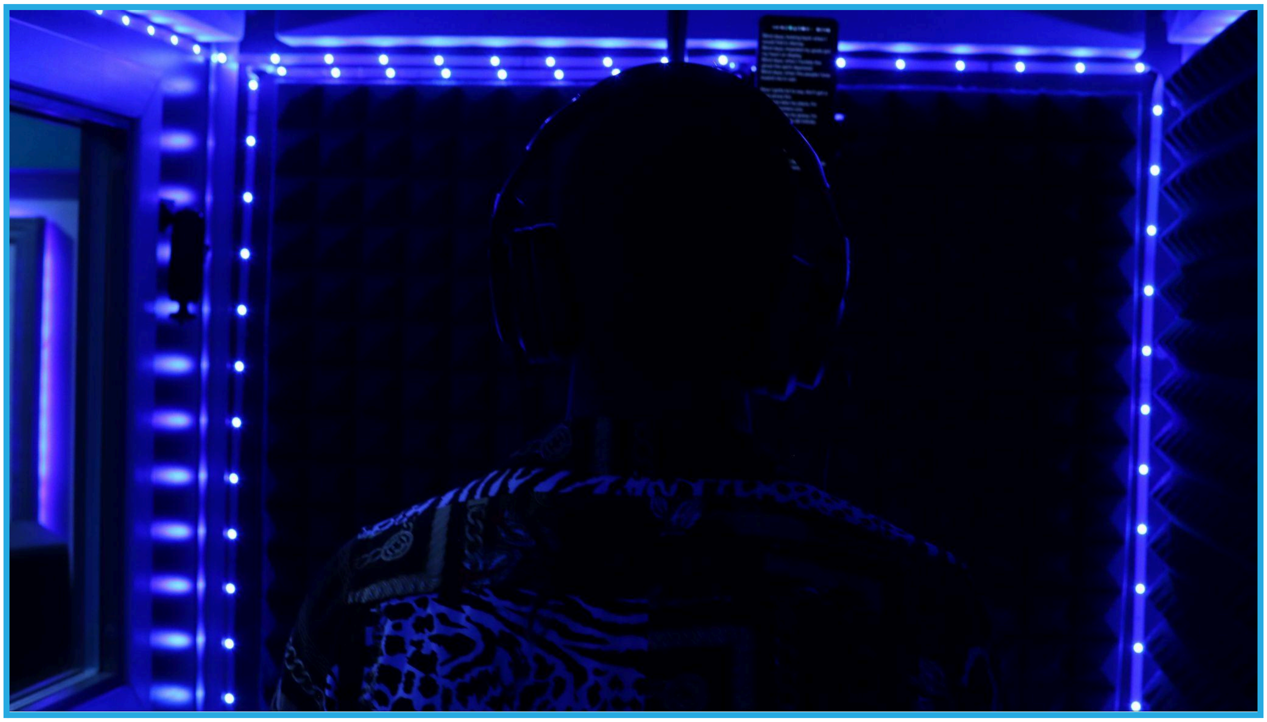
Here they are…

1. Recording
2. Mixing
3. Mastering

Step 1 - Recording

Let’s go over the basic fundamentals required for recording…

First, you must understand that there is a massive difference in the recording process of live instruments and vocals in comparison to recording music digitally via synthesizers, midi controllers and drum machines (used for creating electronic and hip hop music). There’s a lot more involved in the process of live recordings but some basic fundamentals remain consistent in both methods.



Let's look at them separately...

Live recording

Live recording has many more intangibles and many more set up requirements such as:

- Microphones
- Microphone placement
- Cabling
- External outboard gear
- Understanding I/O (inputs and outputs)

Digitally sampled recording

Digitally sampled recording requires much less to get going:

- Midi controller or synthesizer
- Drum machines
- Samplers
- Usb or midi connectors

While both require much different equipment, the **fundamentals of recording** still apply to both methods.

- You still will need a metronome or click track to find a tempo to get started.
- Most people still tend to begin the recording process with drums and bass (otherwise known as the rhythm).
- Then harmonies are added to the mix (piano, guitar, sax etc.).
- Once the foundation for the instrumentation is set, it is time to record the vocals.
- Generally once the vocals are performed, additional instrumentation is added or removed to correctly capture the mood and feeling of the song.

Once the recording process is up to snuff, it is now time to edit and mix it all together.

Step 2 – Mixing

What is mixing?

In a nutshell, the easiest way to describe mixing is; the art of balancing your individual recordings to sound cohesive when combined together into one track. Although mixing is an extremely technical skill it is also a unique art form that requires an immense amount of creativity.

Now there are countless ways to achieve this balance but the fundamental tools used to do so are constant and essential.

The 5 key basic fundamental tools used in the mixing process are as follows:

1. Level adjustment and automation
2. Equalization
3. Dynamic processing
4. Panning
5. Reverberation "reverb"

Another key aspect to mention in the "mix" process is **Editing**.

Some people might add editing as its own separate aspect of the recording process but we include editing as a part of the mixing process. Either way is fine, just understand that the editing process of recording is just as important as any of the other categories.

Let's go cover the EDITING process...

Once your tracks are all recorded, you can guarantee they are going to need to be edited. Very rarely do recordings come out perfect. Even the most highly experienced musicians and recording engineers rarely get perfect recordings.
Editing is definitely not the most glamorous and exciting step of the recording process but is just as integral to a great final product.

There are many things involved in the editing process but **here are some of the key tasks that generally need to be done for every recording:**

1. Labeling
2. Noise reduction
3. Pitch editing
4. Arrangement
5. Comping
6. Time editing

Step 3 - Mastering

Once your recording is edited, mixed and well balanced, it now must be exported into one single stereo file to be mastered. The industry term for this is called "bouncing".

Mastering is essentially putting the final touches or "polish" on the song and formatting it correctly for several avenues of distribution. You want to make sure that your music sounds good on all platforms of distribution and all listening environments. Mastering is what you need to achieve this.

Here are some of the key tasks involved in mastering:

1. Acute Equalization
2. Maximizing volume level
3. Compression
4. Adding harmonics and saturation
5. Stereo widening
6. Dithering
7. Making sure the audio sounds good across all playback systems

Once the final "polish" is applied, the song must then be prepared for the various formats of distribution by being converted to the proper sample rate, bit depth and file format.

For example, the file format, sample rate and bit depth needed for ITunes distribution is going to be different for cd distribution.

Can you master your own music?

That is a question that has been debated for years. Maybe the better question to ask is **should you** master your own music?

Here's my opinion…

There is no doubt that professional mastering is complex and technical by nature and requires very high-end (expensive) monitors to be done well. To be an effective mastering engineer you must have a highly trained ear for sound and frequencies, which takes years to develop. That being said, if your intentions are to be a professional artist and want to be looked at as a professional, we highly recommend outsourcing your mastering needs and leaving it up to a true professional. Believe us, it's well worth the money.

If your intentions are not of a professional magnitude, outsourcing for mastering probably isn't the wisest way to spend your money. Over time as your ears develop and are more trained to understand how frequencies and sound works, then go ahead to start learning and practicing to master your own music.

AI Mastering Software

These days, there are quite a few AI Mastering platforms such as landr that use AI technology to master your audio in just a few moments. And to be honest, when they first came on the scene I wasn't a huge fan. I thought they were limited in their capabilities to produce a good master for every type of song. But over time as they made refinements to the technology, I actually see it as a viable option depending on your situation. If you're new to music and want to release your music on streaming platforms and don't know too many people that can master your music, then these AI services are definitely better than having no mastering done to your tracks at all. Now this option is not as good as having an experienced mastering engineer do the job but it can definitely be sufficient if you are releasing one song at a time.

2. Signal Flow and Gain Staging

There are two important aspects of the recording and mixing process that are often very misunderstood and overlooked by people new to recording:

- Signal flow
- Gain staging

Signal flow is the path that the audio signal takes from the input to the output in an audio console. The signal first enters the console through an input. An input is usually a balanced signal that enters at mic level.

Gain staging is the process of managing the relative levels in a series of gain stages throughout the signal chain to prevent introduction of noise, distortion, and clipping.

Ideal gain staging occurs when each component in an audio signal flow is receiving and transmitting signal in the optimum region of its dynamic range. In layman's terms don't clip while dialing in each of your levels starting from the first initial piece of hardware in your signal chain.

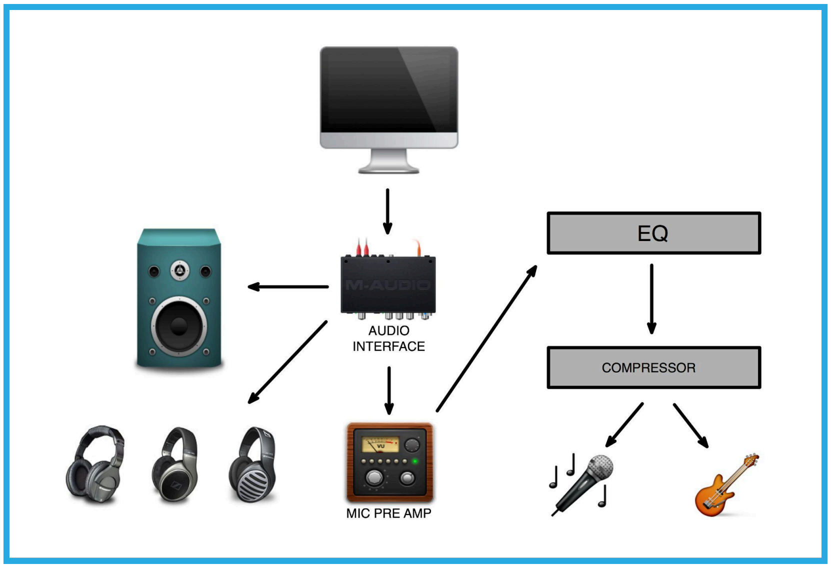
So what does all of that mean?

When we first begin recording, it seems like human instinct to want to record everything as loud as possible. The fact of the matter is that **recording "hot"** (at high levels) is probably the worst philosophy you can have when recording.

This belief really just stems from lack of education and experience. The reality is that you should only be focused on not clipping and distorting the signal no matter how low the recording seems. It is not very difficult to get the level back in the mixing process once you know what you're doing. Making sure the levels aren't clipping in any stage of the signal chain is paramount. Loudness should not be a concern till the mixing and mastering stage.

What is a signal chain?

The signal chain is the signal processing chain of all electronic components (inputs) that an audio signal passes through before it reaches the final output stage. A simple diagram can best explain this:



Understanding gain staging and signal flow takes practice and getting to know your gear, so don't be discouraged if you can't seem to get it right at first. **"Stick with it and don't clip it"** is a good way to approach it.

3. Cables

Some of the best advice we can give when it comes to cables is to never underestimate their value and importance, they are essential to have your studio running at an optimum level. We briefly discussed in the second chapter that you should not cheap out on cables for many reasons. Although it is tempting when the budget is tight, you will be glad you shelled out the extra dollars.

We're stressing this because we had to learn the hard way for years before we understood this. It seemed like we needed to get a new cable every other week because they just kept dying out on us. Not only was this a huge waste of time to keep going back to the store (many times in the middle of sessions) to get new ones, we ended up spending the same amount of money replacing the cheap ones in the long run anyway.

In this chapter we will discuss the different types of cables used and the roles they play in a recording studio:

1. Analog and digital cables
2. Balanced and unbalanced cables
3. The different types of audio connector cables

Analog and digital cables

There are two types of cables used in recording studios today.

- **Analog cables**
- **Digital cables**

The difference between the two is that analog cables transfer through an electrical signal via wires and digital cables transfer information via 1's and 0's (binary code).

The electrical nature of analog cables often means that they can add more noise to the signal (especially in cheaper made products). Digital cables on the other hand tend to have less noise (**signal to noise ratio**) introduced to the signal.

But that is not to say all analog cables are going to introduce noise to the signal, it just means that it's more likely than with digital cables.

Analog cables come in two primary forms:

- Balanced
- Unbalanced

Balanced: Balanced cables have insulation protecting the wires from external electrical and radio interference.

Unbalanced: Do not have insulation protecting the wires from external electrical and radio interference leaving the signal more vulnerable.

Balanced cables tend to have less level and a better s/n (signal to noise) ratio. They also have a slightly lower frequency response in higher frequencies.

There are 3 types of cables you may need for connecting your audio source:

- **Line level** – requires a balanced cable connection.
- **Instrument level** – requires an unbalanced cable connection for line level via D.I. box or preamp.
- **Mic level** – requires a balanced cable connection for line level via microphone pre amplifier.

The different types of audio connector cables

Most of your audio products are going to be connected by three main types of connector cables:

- XLR
- Trs
- Ts

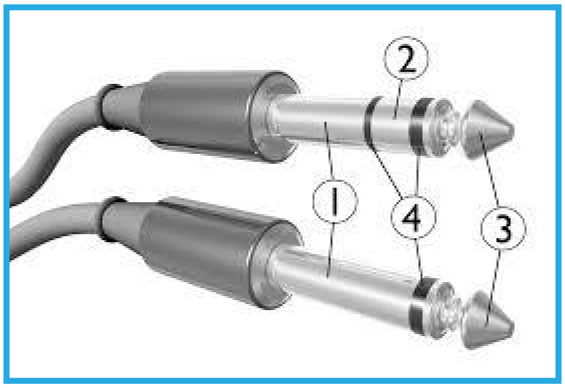
XLR – a standard XLR **male** connector has 3 pins, while the **female** connector has 3 holes. So it should be easy to remember which one is which. Balanced 3 pins XLR connectors each have their own individual purpose:

1. Positive
2. Negative
3. Ground

Trs (Tip ring sleeve) – is a balanced quarter inch connector that has the same technical make up of xlr connectors.

- Tip = positive
- Ring = negative
- Sleeve = ground

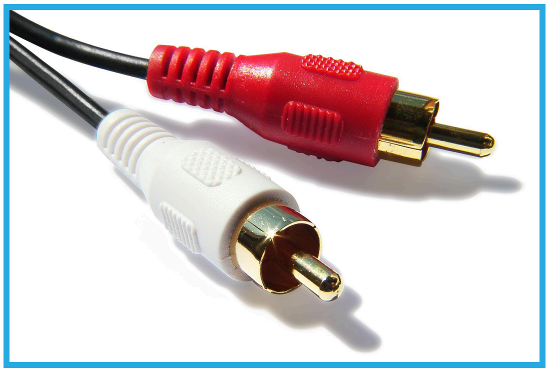
Ts (tip sleeve) – is an unbalanced quarter inch connector



- Tip = positive
- Sleeve = ground

Most microphone connections require some form of XLR connection while quarter inch cables are typically used for instruments such as keyboards, guitars and bass. Most newer studio monitors on the market give the option to be connected with both types of cables.

RCA Cables



In general RCA cables are identified by their red and white heads and 8 mm connectors. RCA cables are generally not used in a professional audio capacity because they don't match as many impedance requirements and they don't carry shielding through the way quarter inch and XLR connectors do. RCA cables are used mainly for TV, video and consumer audio.

Midi Cables

Midi cables are used to transfer midi information from source to another. Midi cables have 16 paths to transfer information through, which is why midi controllers can control so many parameters.
They are 3 types of midi cables used in audio:

1. **Midi in** – receives midi information.
2. **Midi out** – transmits midi information.
3. **Midi through** – receives and transmits information.

Although midi cables were specifically designed to transfer midi information from the electronic source to a digital source, usb cables also have the capability of transmitting midi data and are becoming much more popular connectors for midi controllers. It is said that usb is up to 15,000 times faster than midi cables but are known to have more latency and jitter.

Optical Cables

Optical cables are digital cables used to transfer multiple channels of data through one cable.

There are 2 optical cables used in pro audio:

- Adat
- Spdif

Adat

Adat cables are digital cables that transfer digital audio, which is beneficial for connecting external components to one another.

For example, if you are using an audio interface that only has two mic preamps and you want to start recording drums and need more inputs. If your interface has an adat connection you can connect an external mic preamp with up to eight channels if needs be and run it directly through your interface. This saves the money and hassle of purchasing a new interface with more inputs.

Spdif

Spdif cables are very similar to adat cables and are used in the same manner. The difference is that spdif cables can only carry two channels of data as opposed to the eight channels of adat cables.

D Sub Connector Cables

D sub cables are 25 pin connectors and are most commonly used to connect multiple inputs and outputs to a single ended connection point. D sub cables are also used to "daisy chain" multiple external pieces of hardware.

Power Cables

Power cables provide the electrical signal to power the hardware unit.
There are two types of power cable connections: Balanced and Unbalanced.

Balanced: Most audio recording hardware units require a balanced connection to reduce the possibility of output current bursts and reject electrical and radio signal interference.

Unbalanced: Typically unprofessional hardware is using an unbalanced power connection.

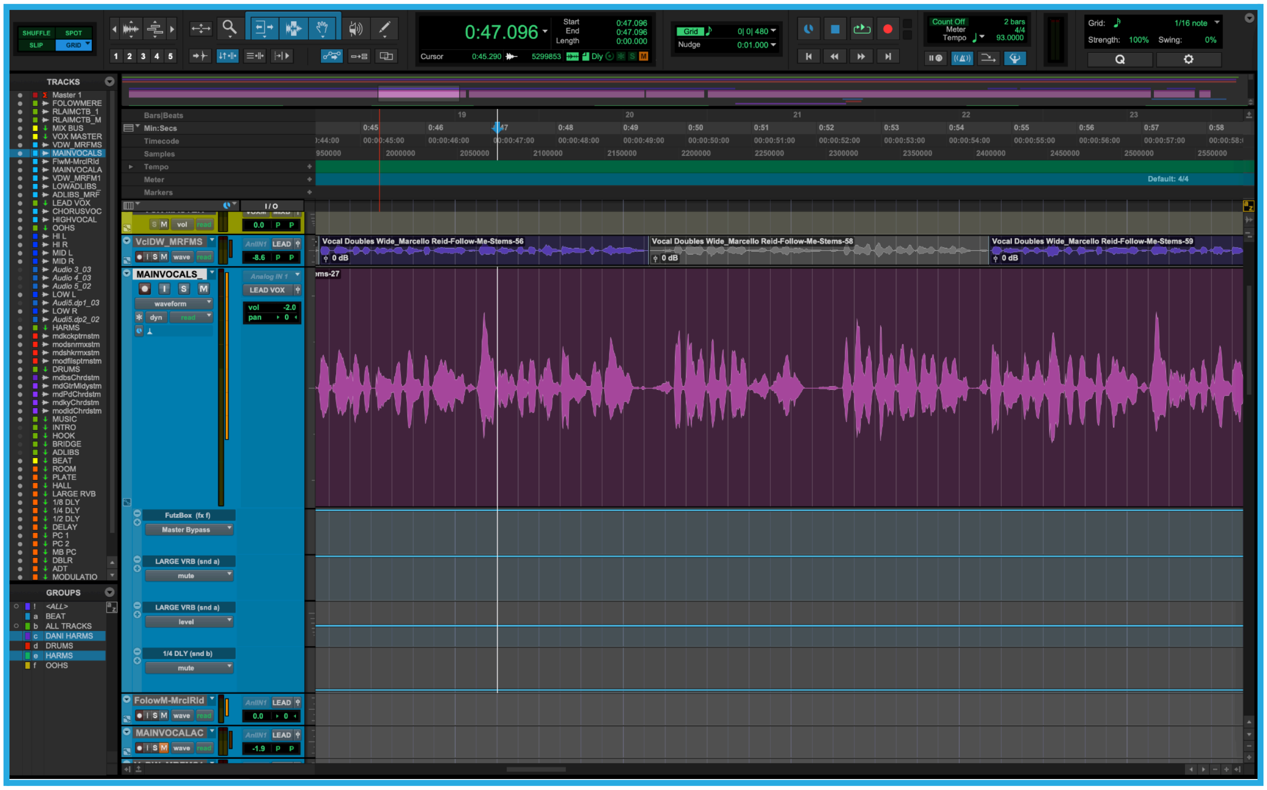
Final thoughts on cables

Like most products, there are different levels of quality available to us and in many cases we get what we pay for but of course that doesn't mean that all expensive products are the best either. In our experience, when it comes to cables you get what you pay for the majority of the time but if you can only afford cheaper cables here's what you'll be faced with:

Why cheap cables suck!

- Cheaper wire tends to break easier and is not as conductive.
- Cheaper cables have lower quality soldering, which makes the connection more susceptible to breakage.
- Cheaper cables tend to use lower quality materials, this affects the radio and electrical interference. Meaning more possibility of having interference issues.
- Cheaper cables have less quality connectors which tend to break and alter conductivity.

4. Digital Audio



The rise of the home studio era is that of legend. The music products industry is now a multi billion-dollar industry that generates more revenue than global music sales. So as the music industry continues to struggle, the home studio and music products industry continues to grow year after year. This growth and rise of a new era in recording is largely attributed to the development of digital audio software and recording products.

No longer do we need to spend tens of thousands of dollars on big analog consoles and professional engineers that in the past, only the major record labels and wealthy people had access to. Not only were the analog consoles incredibly expensive, they are also incredibly inefficient. By inefficient, we mean not having the ability to edit and record as many takes as you want with the press of a button. In the analog world, this is an extremely difficult and time-consuming task.

On the flip side, there is digital audio software available that can be downloaded for free and can essentially do what a $100,000 console can do but much more efficiently. This affordability and efficiency allows any one with access to a computer and a microphone the ability to record music anywhere. Which is truly amazing when you stop to think about it. Not only has digital audio become the standard for home studio enthusiasts but it has now become the standard in commercial studios all around the world.

For most of you, your journey of recording is going to begin in the digital realm. Meaning you will be recording your music through a computer on recording software. This software is referred to as a digital audio workstation (DAW).

But before we can discuss "DAWS", **there are 10 topics that you need to understand when it comes to digital audio…**

1. **Audio interfaces** – refer back to **Module 1** for information on audio interfaces as we have already covered what they are, how they work, and how they apply to digital audio.
2. **Digital converters**
3. **Sample rate**
4. **Bit depth**
5. **Buffer sizes**
6. **Latency**
7. **Master clocks**
8. **Audio file formats**
9. **File organization**
10. **DSP (Digital Signal Processor)**

We mentioned at the beginning of the guide that we are going to try to avoid getting too technical with any of this stuff but many of the topics involved with digital audio are technical by nature and must be explained as such. So if you're not a computer person, we know some things will be confusing but just follow along the best you can. Unfortunately this stuff cannot be avoided and is a major aspect to the recording process. In time these things will begin to make much more sense to you just like everything else in the recording process.

2. Digital converters

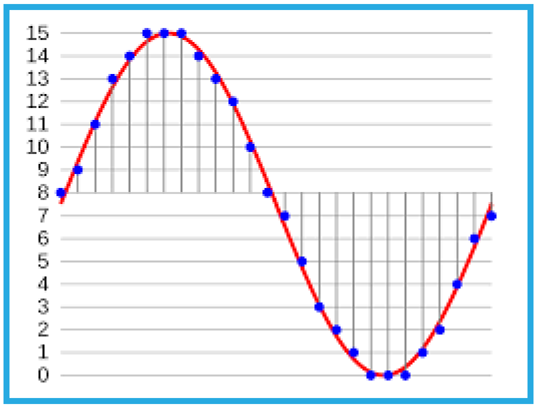
According to Wikipedia - a digital to analog converter is a function that converts digital data (usually binary) into an analog signal (current, voltage, or electric charge). An analog-to-digital converter (ADC) performs the reverse function. Unlike analog signals, digital data can be transmitted, manipulated, and stored without degradation, albeit with more complex equipment.

Digital converters will generally come in 2 forms:

1. In an audio interface
2. As a standalone piece of hardware

For home studio's, audio interfaces will most likely be your source for a digital converter. These converters essentially allow the audio to come in and out of your computer at a certain rate of speed and level of sound quality.

3. Sample Rate



The sound quality you receive from your converters is largely going to be determined by sample rate.

So what is sample rate?

Unfortunately there's no easy way to explain sample rate, so bear with us as we are going to have to get a little technical here.

The **sample rate** is the number of samples of a sound that are taken per second to represent the event digitally.

The more samples taken per second, the more accurate the digital representation of the sound can be. For example, the current sample rate for CD-quality audio is 44,100 samples per second. This sample rate can accurately reproduce the audio frequencies up to 20,500 hertz, covering the full range of human hearing.

When it is necessary to capture audio covering the entire 20–20,000 Hz range of human hearing, such as when recording music or many types of acoustic events, audio waveforms are typically sampled at 44.1 kHz (CD), 48 kHz, 88.2 kHz, or 96 kHz. The approximate double-rate requirement is a consequence of the Nyquist theorem.

Sampling rates higher than about 50 kHz to 60 kHz cannot supply more usable information for human listeners. Early professional audio equipment manufacturers chose sampling rates in the region of 50 kHz for this reason.

There has been an industry trend towards sampling rates well beyond the basic requirements: such as 96 kHz and even 192 kHz. This is in contrast with laboratory experiments, which have failed to show that ultrasonic frequencies are audible to human observers.

One advantage of higher sampling rates is that they can relax the low-pass filter design requirements for ADC's and DAC's (analog to digital converters and digital to analog converters). That is the science of sample rates but don't get too caught up in it right now. Just keep this in mind and you should be fine. The Audio Engineering Society recommends 48 kHz sampling rate for most applications but gives recognition to 44.1 kHz for Compact Disc and other consumer uses, 32 kHz for transmission-related applications, and 96 kHz for higher bandwidth or relaxed anti-aliasing filtering.

4. Bit Depth

In digital audio, bit depth describes the potential accuracy of a particular piece of hardware or software that processes audio data. In general, the more bits that are available, the more accurate the resulting output from the data being processed.
Bit depth is used in analog-to-digital converters and digital-to-analog converters when reading plug-ins, and when recording audio via a digital audio workstation (DAW)

Bit depth is the number of bits (binary digits) you have to describe something. Each additional bit in a binary number doubles the number of possibilities.

Here's how the math works...

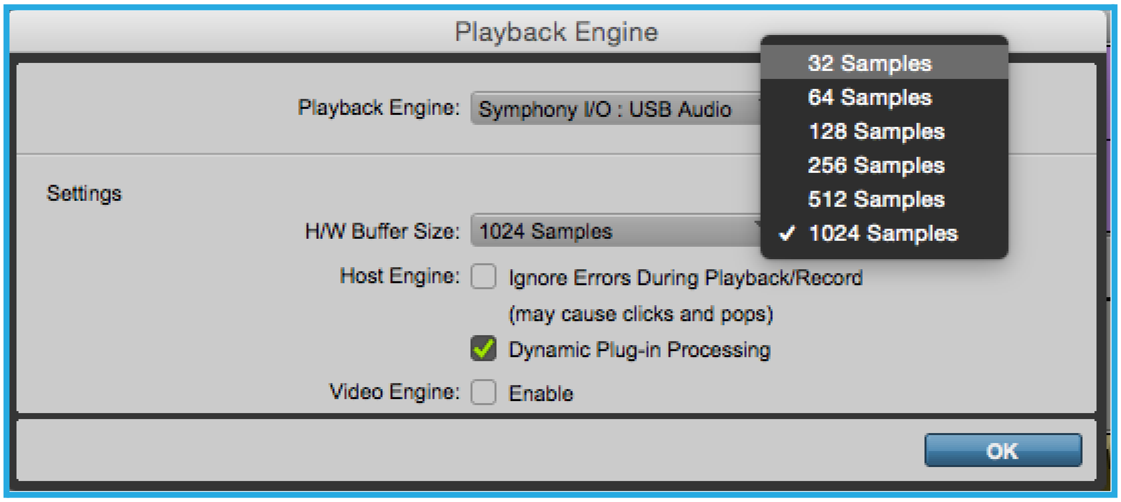
- A 4 bit sequence = 16 possible levels
- An 8 bit sequence = 256 possible levels
- A 16 bit sequence = 16,536 possible levels
- A 24 bit sequence = 16,777,216 possible levels

One thing to remember is that the greater the bit depth, the more dynamic range of the audio waveform resulting in more headroom. And trust us, there can never be enough headroom in a recording.

Headroom - In digital and analog audio, headroom refers to the amount by which the signal-handling capabilities of an audio system exceed a designated level known as Permitted Maximum Level (PML).

5. Buffer Size

To our ears when we playback audio from our computers, from the time we press play to the time the audio is playing back from our speakers appears to be instantaneous. But it's actually not. Technically there is a micro delay. The time it takes for your soundcard to process the audio information through the speakers is referred to as the **buffer size**.



Recording software will generally give you the option of 5 standard buffer sizes and the quality of your sound card and converters will determine how effective those buffer sizes will be.

The 5 standard buffer sizes:

1. 64
2. 128
3. 256
4. 512
5. 1024

The buffer size also works in relation with the sample rate.

Here's how…

The buffer size can remain at a fixed value regardless of what the sample rate is set at. Meaning the higher the sample rate, the quicker audio will still pass through the buffer and the lower the latency will be.

To keep it simple and less complicated, the smaller the buffer size the quicker audio will pass through. The larger the buffer size the more CPU power you will gain. One thing to keep in mind with a small buffer size is that if your computer can't handle the low processing you will get dropouts and artifacts in your audio when played back. Just set the buffer size higher if these problems occur.

In the world of recording, a safe way to approach buffer sizes is to record with a smaller buffer size (hence the low latency) and mix on a higher buffer setting (hence the much needed cpu power required when mixing large sessions).

6. Latency

Latency has always been an issue with digital recording because of all of the factors involved with the technicalities of how audio is transmitted through computers. But as technology evolves, the issue of latency is becoming less and less of a problem. The converters being made now are incredibly transparent with low latency and of course, computers are getting faster by the day and are able to process data so much faster. The mathematical equation to figure out how much latency you will get is fairly simple:

Number of samples divided by sample rate = latency (in milliseconds)

- Ie – 512 samples divided by 48 khz = 10.6 ms

Let's address the factors that contribute to latency in digital audio:

- Digital to analog converters
- Analog to digital converters
- Buffer size
- Processing effects and plugins

Most converters on the market today aren't going to give you much latency issues. It is said that converters will only give up to a maximum of 5 milliseconds. (The better the converter quality the less latency). Also the better the converters, the better quality your sound will be. That is one of the primary reasons why you pay for expensive audio interfaces. Quality converters are not cheap! But in my opinion it is worth every penny when you want to start getting a more professional sound from your home studio. Cabling also plays a factor here as well. Most new audio interfaces on the market today are usb c and thunderbolt. Which is currently the best connector type as far as speed and quality. Older usb 2 & 3 cables are not going to be as powerful when it comes to conversion quality and speed.

Buffer size also plays a massive role in latency so remember, keep the buffer size low for less latency when recording.

7. Master Clock

A master clock is a clock that provides timing signals to synchronize slave clocks as part of a clock network. It is essential for MIDI connections when timing is a critical factor for recording one device to another.

Effects and Plugins

If you are working in a recording session that has a lot of plugins and effects inserted, they will eat up your CPU in a hurry and bring your session to a crawling halt if your computer isn't primed for pro audio. In that case increase the buffer size accordingly until the delay is no longer present. If that still doesn't work you will either have to deactivate some plugins or, bounce out the processed tracks you have completed into one stereo file and continue processing the remaining tracks.

Keep in mind that different plugins eat up more CPU than others. So check the specs on your plugins and mix accordingly. Some plugins are notorious CPU hogs!

8. Audio File Formats

There are a number of audio file formats out there in the digital world, but there are two formats in particular that dominate the pro audio and music domain.

- MPEG Layer-3 files or more commonly known as **MP3**
- Wave files – commonly known as **WAV**

WAV Files – Without getting too technical, .wav files are an uncompressed audio file which delivers better sound quality. The con of uncompressed files is that they are larger in size, which can make them more difficult to transfer through certain transfer mediums such as standard email. Wav files are referred to as "**uncompressed**" audio formats.

MP3 Files – Mp3 files are by far the most popular file formats used for music and audio. Mp3's have become the standard for the playback and transfer of music for most digital audio players. Mp3 files are referred to as a "**lossy compression**" audio formats which means that the audio is compressed making the file smaller and easier to transfer. Of course with a compressed audio file, the sound quality will be compromised to a larger degree.

3 other file formats worth mentioning are:

- **AIFF** (audio interchange file format)
- **AAC** (advanced audio coding)
- **WMA** (windows media audio)

In brief, **AIFF** files are an apple based uncompressed file format that is now compatible across many applications and audio players. Some say it is apple's version of a wav file.

AAC - files have quickly become an audio encoding standard for lossy audio compression formats. Reason being, it has been designed to produce slightly better sound than an mp3 at the same bit rates.

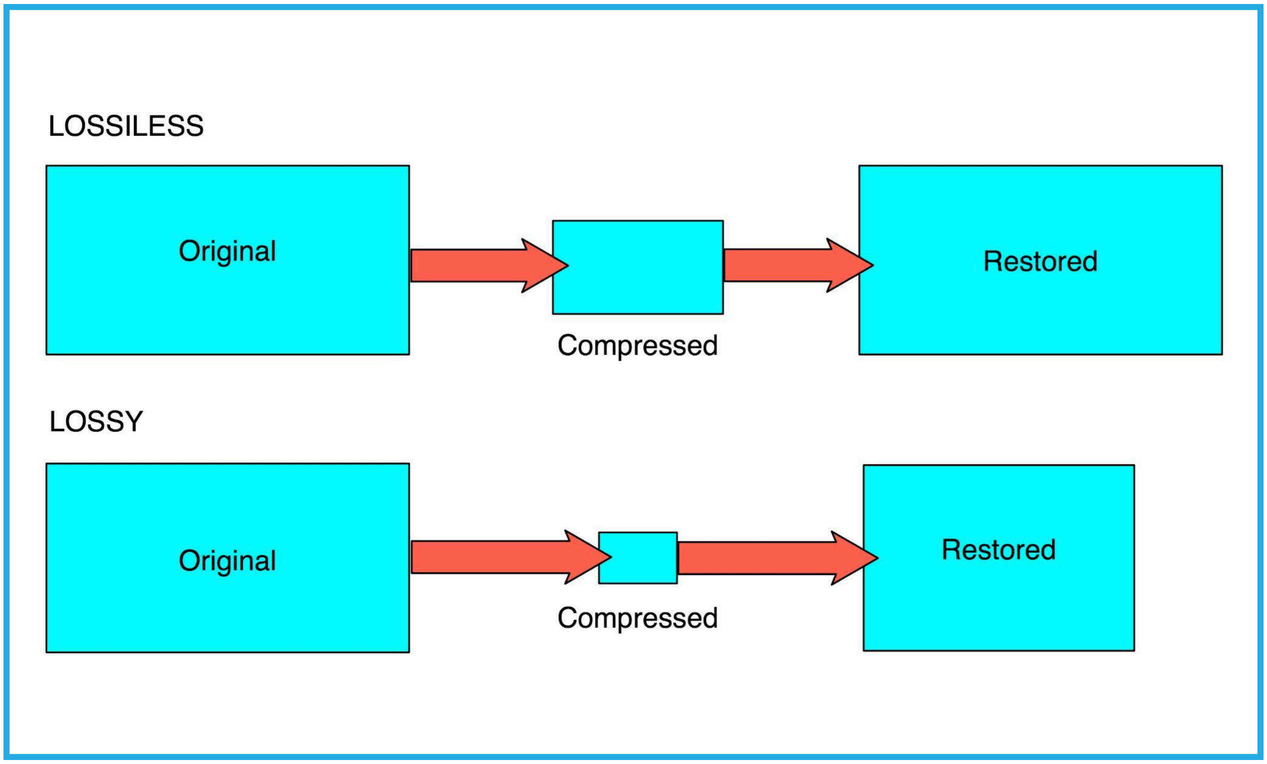
WMA - files are a windows based file format, they come in lossy and lossless compression audio formats. WMA files were originally created to be a competitor of the mp3 file format and are fairly popular among windows users.

Codec - The way audio is compressed (or uncompressed) is referred to as the codec, which determines how small the file size can be while retaining the sound quality of the file.

What is an audio format?

All of the file formats out there fall under one of the three audio format categories, which are:

1. **Uncompressed Audio Formats**
2. **Lossy Compression**
3. **Lossiless Compression**



Uncompressed Audio Formats are simply that. Uncompressed audio files that hold all of the data available but result in larger file sizes.

Lossy Compression applies a compression algorithm to an uncompressed file to eliminate what it perceives to be unnecessary or irrelevant information. This removal of information results in worse sound quality but much smaller files making it much easier to transfer over the internet.

Lossiless Compression applies compression to a file but doesn't eliminate any information, maintaining the sound quality of the file more so than a lossy compression format.

Can you convert between audio formats?

Of course you can. There's a ton of software out there that allows you to convert audio formats when needed. Just keep in mind that when converting from a lossy or lossiless file to an uncompressed file such as a wav file will not improve the sound quality of a compressed file. All it does is simply change the file format, therefore making it unnecessary and almost pointless.

Conversely, if you wish to convert an uncompressed file to a compressed file, by nature, the compression that takes place will reduce the sound quality during the conversion process.

9. File Organization

Now if you are a regular user of computer software, surely you understand the importance of keeping your files organized, especially if you are a professional in some capacity.

Well, when it comes to recording, file organization can never be stressed enough. Developing great organizational habits from the beginning will be a decision and habit that will save you an unbelievable amount of time and stress in the long run.

So keep your files as organized as possible!

Since all recording software has their own unique file management systems, we will cover the file management fundamentals that apply to pretty much all software. Here are some habits you should implement from the beginning…

Good file organizational habits are:

- **Folder management** – make sure you always name and label you folders accordingly and and always create more specific sub folders when needed. Don't overfill your folders!
- Develop a consistent routine of **cleaning up your desktop**. Schedule a specific time frame to do so. (Example: every two weeks clean your desktop)
- Have a **dedicated external hard** drive to back up your work onto and back it up regularly.

10. Digital Signal Processors (dsp)

A **digital signal processor** (dsp) is a specialized processor that is designed to optimize the fundamental needs of digital signal processing. Digital Signal Processors process data in real time, making it ideal for applications (such as digital audio) that cannot tolerate delays. Digital signal processors take a digital signal and process it to improve the signal making clearer sound, faster data or sharper images. So in the audio world, an analog to digital converter will take the analog signal and turn it into the digital format of 1's and 0's (binary code). From here, the dsp takes over by capturing the digitized information and processing it. It then feeds the digitized information back for use in the real world. All of this is done at lightning speeds.

So how does this affect you and your recordings?

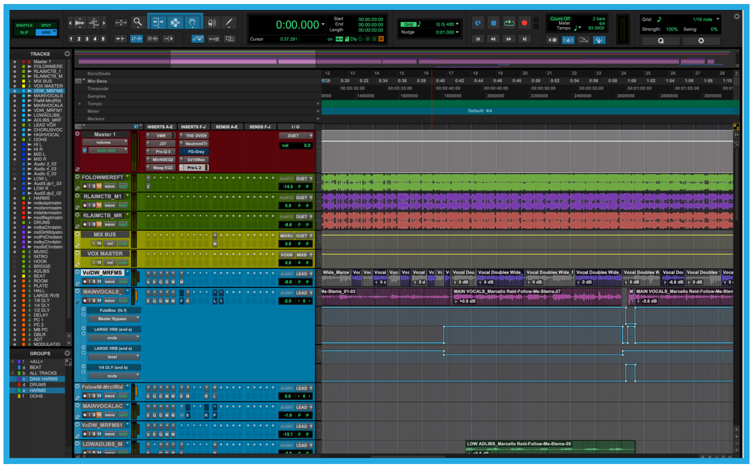
Actually it's very simple. The more information being sent through the dsp, the slower the dsp will process the signal. So if you are working in your DAW and you keep adding more and more effects to your tracks, each effect then adds more for the dsp to process in turn slowing down the audio that is being played back.

5. Recording Software

First a little history...

In 2016, recording software and computers now dominate the recording industry. Even the most "old school" of professional engineers that only knew "analog recording" are giving up on the debate and are primarily recording and mixing "in the box" now. So why have they finally decided to give up the debate?

Well nobody could ever deny the efficiency and editing capabilities of recording software. That wasn't really the debate. What took the extra 15 years of resistance was the difference in sound quality. There is no mistaking the rich tonal qualities and vibe of analog recordings in comparison to the almost too clean "vibeless" sound of digital recordings.



There is no doubt that characteristics and subtle artifacts that are inherent with analog circuitry bring more to the table than the inherent nature of the circuitry of digital software from an audio and musical standpoint. Because of this noticeable difference in sound quality (which is still actually opinion based and not scientific fact), engineers kept resisting to jump on the digital train.

Software companies quickly took notice of this issue and began developing products that emulated this "analog" sound. This was great for digital people and analog people. The problem was that it just didn't quite sound the same as the real thing. So in theory the problem was solved, the technology just needed to be refined and perfected.

In time, technology advanced and software companies developed better techniques in creating more accurate algorithms so the plugins sound just like the original piece of hardware and analog consoles. Now the accuracy of these algorithms is still highly debated in the audio world but for most top professionals, they are accurate enough to make them finally crossover to digital software.

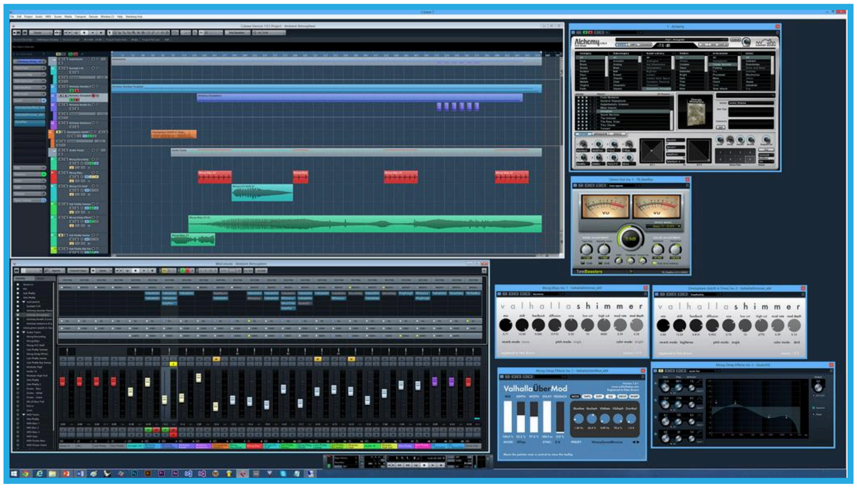
When first learning how to record, it is difficult to wrap your head around all of the software used in digital recording.

So we will break down the basics of the most used and most important software used today:

- DAW (Digital Audio Workstation)
- Virtual instruments

- Plugins
- The 3 most important mixing processors
- Other key effects processors

6. DAW (Digital Audio Workstation)



Like we discussed in the first module, there are plenty of DAWs on the market and they all range in prices and functionality. But the fundamentals of all DAW software are essentially the same.

Here is the list again for a quick reference on popular DAWS used today:

1. ProTools by Avid (Mac & PC) $$$
2. Logic Pro X by Apple (Mac) $
3. Live by Ableton (Mac & PC) $$$
4. Cubase by Steinberg (Mac & PC) $$
5. Fruity Loops by Image Line (Mac & PC) $
6. Reason by Propellerheads (Mac & PC) $$
7. Studio One by Presonus (Mac & PC) $$
8. Sonar by Cakewalk (PC) $

Pick one that suits your budget and check out our **resource center** for websites that offer great online tutorials to help get you started!

Virtual Instruments

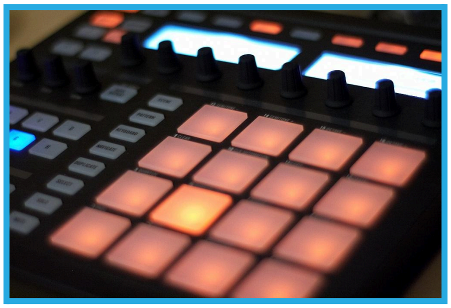
Virtual instruments are just that, virtual instruments. Software companies now allow us the ability to play instruments without physically needing a real instrument. Every sound imaginable is available to us through virtual instrument packages and if we can't find the sound we want, we can just create them with virtual synthesizers.

With virtual instruments, your creativity for music production is literally endless. But just like any other emulation software, there are debates on the accuracy of sound when it comes to certain instruments. We will say this though, the top companies that make virtual instruments put a lot of time, money and effort into getting they're sounds accurate and in our opinion do a great job emulating instruments and sounds.

The problem producers have with virtual instruments is that it's very difficult to play a complex guitar or bass riff on a keyboard. So while they can be very effective, they still have their setbacks.

Virtual instruments can be triggered many different ways:

- Your computer keyboard
- Midi controller
- Synthesizer
- Drum machines
- DAW controllers



As to where to get them, most DAWS will come stock with virtual instruments for you to play around with. But there are literally hundreds of companies that produce virtual instrument software, which makes it very difficult to choose.

Here is a list of a few goodies to start with and get you going:

- **Native Instruments**
- **Sample Tank**
- **samplitube**
- **Toontrack**

Plugins

Audio plugins are composed of effects and virtual instruments that can be inserted onto your tracks in your DAW. Plugins are a vital piece to the recording and mixing puzzle. Because of this, all DAWS come stock with their own set of plugins that the companies design themselves.

Thanks to high-powered computers and plugins, we can now add as many effects as we wish to our songs with just a click of a button. This is a feat that could not be accomplished in the analog world, just another reason why "old School" engineers continue to hop on the digital train.

As people get more experienced with recording and mixing, they eventually venture out into the vast universe of third party plugins. By "third party" we mean plugin software that does not come stock with your DAW. Now when I say vast, I mean VAST!

Because of the explosion of digital music, it seems a new plugin company comes out every other week to try and get a piece of the pie. Definitely can't blame them though. A lot of the trial and error in the algorithm process has been done already. It's more about design and functionality nowadays because the analog technology is readily available, which makes developing the plugins a little easier (but we are not saying it's easy by any means).

There are 3 main plugin processors used in the mixing and recording process:

1. Compressors
2. Equalizers
3. Reverbs

To be honest, it can take years before one can learn to use these processors effectively and there is a lot that goes into how to use them, which is far beyond the scope of this guide. So for now I will just explain what they do in the simplest form.

Compression

Compression is used to lessen the dynamic range of the softest and loudest volume levels of an audio signal. Compressors take the highest peaks and bring the volume level down and take the lowest parts of the signal and raise the volume up to essentially create an even sounding piece of audio.And keep in mind, we don’t want to flatten the dynamic range of our recordings, we want to use compression to control the dynamic range of our recordings. When vocals and instruments are recorded, by nature, the level of the audio recorded will be high in some parts and lower in others. Although compression has many functions, the most important function it performs is turning the high points and low points of a recording into an even performance, making it much more pleasing to the listeners ears. That is probably the most non-technical description we could give without confusing you too much on the subject.

Equalization

When I first explain equalizers to my students, I always refer to them in terms of bass and treble as most people can identify with those terms and are familiar with what they are and what they do to sound. Over the years I’ve found that the simplest explanation of an equalizer (eq) is - a tool used to adjust the volume of specific frequencies in an audio signal. Meaning it's used to adjust the amount of bass or treble in a signal. And that really is essentially what an eq does for us on the most fundamental level.

Equalization is a technical way to get all of the instruments and vocals in a track to blend cohesively in a manner that allows you to hear all of the instruments and vocals clearly. Certain instruments have the same frequency range and therefore compete to be heard when mixed on top of each other. Equalization (eq) is the way to add or remove the frequencies in order for the finished product to have all of the instruments audible and sounding great.

Reverb

Reverb is created when sound sources of any kind reflect off the walls of a room or space causing multiple reflections to build up and then decay as the sound is eventually absorbed by the space and the objects in the space. More Simply put, reverb is a sound that any given space creates from the reflections of any given sound source.

Reverb is essentially the echo you hear when in a large hall, church or room. Of course that is an extreme case of reverb but you get the point. The truth is, there is reverb in all spaces, but the less sound has to travel the less reverb you will hear. Reverb is important because it gives your recordings a sense of space that would otherwise have your recordings sounding unnatural without it. We generally tend to record in acoustically treated spaces eliminating most of the natural sound reflections where reverb comes from. Since acoustic treatment is so important in so many recording scenarios, adding that sense of space to a dry recording with a reverb plugin is almost a necessity.

The beauty with reverb technology in plugins is that you can emulate whatever space you desire. From Wembley stadium, to a jazz club, to a bedroom and everything in between, it’s all there for your creative use. While in the old days, you would have to settle for the sound of the space that you recorded in the period.

Other Key Effects Processors

Delay

A delay occurs when an input audio signal is recorded and sent to an output or audio storage medium and is then sent back to the input signal after a specified period of time. This time it takes for the audio signal to be sent back to the input is what is called delay. When applied in music, delays are set to milliseconds and seconds. Anything longer than a couple seconds becomes non musical and non pleasing to the listener.

Delays have many uses and functions in the mixing process. Similar to reverb, delays can also give a sense of space to a mix when applied properly. If you are new to recording you probably refer to delays as echoes. But don't worry, it's all the same thing and a lot of plugin companies use the term echo as well.
Strategically placing delays on specific parts of a vocal or instrument can really add some width, character and feel to track which is why they are used so readily in music today.

Stereo Imagery

According to wikipedia, stereo imaging refers to the aspect of sound recording and reproduction of stereophonic sound concerning the perceived spatial locations of the sound source, both laterally and in depth. This means that there is a huge stereo field we have to play with and use to place and space our tracks.

So you're probably wondering why you would need to manipulate the stereo field of your music?

Stereo imaging allows for spatial placement of instruments and sounds both laterally (left to right) and depth wise (front to back) within the stereo field. If done well, this placement of instruments in the sound field can make a huge difference in how the song or music is perceived because it gives each piece of the song a sense of its own individual space. This makes it appear more open and make all of the instruments more audible. Advanced techniques in stereo imaging can be used to create mood and feeling as well.

Modulation

In music, modulation is most commonly the act or process of changing from one key (tonic, or tonal center) to another. This may or may not be accompanied by a change in key signature. This changing of keys can be done in a variety of ways that produce some unique effects widely used in recording.

3 popular modulation effects often used are:

- Chorus
- Phaser
- Flanger

Other modulation effects used are **vibrato**, **tremolo and octave** dividers. While all of the effects sound different they are all using modulation to alter the sound source. In most cases modulation effects aren't very musical if exaggerated, the key to using modulation in a pleasing and creative way is subtle adjustments.

Pitch Correction

Pitch correction is one of the most commonly used tools in modern music today and is used in all forms of pop music across all genres and is one of the most important elements of vocal processing and vocal editing. But what usually first comes to mind for people new to music production is that pitch correction is just a vocal effect (autotune) when in actuality it's really a secondary function of the tool. The primary use for pitch correction is to correct the original pitch of a vocal to fit the key of the song as vocalists land flat or sharp on wrong notes. Most singers and vocalists rarely ever record performances in perfect pitch. Even the most seasoned professionals struggle to stay in key all of the time which is why pitch correction software has been so revolutionary for not only the recording industry but the live performance industry as well.

When people think of pitch correction software, the first one that comes to mind is **Autotune**. Surely you've heard it used on many songs before. It's famous for its distinct unmistakable pitch control sound. And while Autotune may be the most well known of the pitch correction software, there are many others out there that are just as good and just as effective.

Saturation and Distortion

Like the rest of the effects processors we've discussed, there is a lot to be explained when it comes to saturation and distortion, so we will keep it simple and brief.

Saturation is a term that describes the characteristics of sound that are produced from analog equipment such as tapes, tubes, and transistors For a long time, engineers in the analog days would record almost everything to tape, in turn having a "saturated" pleasing sound, a sound that could not be found in the digital world for a long time. We have discussed earlier in the section why digital music wasn't so popular with the older generation of engineers. The lack of saturation was a major reason.

Saturation and the right amount of distortion can sound very pleasing to the ears and add "warmth" (an industry term used to describe the sound of analog saturation) to an audio signal that is lacking in digital recordings. This "warmth" was so sought after for so many engineers that plugin companies had to do something about it. Saturation and distortion plugins are amongst the most popular now because of the demand for "warmth".

7. Recording Vocals

In most music, the vocals are what really drives the listeners interest and are often considered to be the most important element of the song, depending on the genre of course. So there's no doubt that great care and technique need to be applied to the recording process to optimize vocal performances.

A lot of people new to recording have the misconception that you need an expensive microphone to achieve great recordings. This is nonsense and couldn't be further from the truth. That is just an excuse, not a fact.

The truth is that you can get great mics for just a few hundred dollars!

It's not the mic that will make or break your recordings; it's lack of experience and knowing what problems to avoid that ruin the quality of your recordings.

Although recording great vocals consistently takes practice and patience, if you can avoid the common issues that can ruin your recordings, technique won't play as much of a factor as you would think. There is nothing more depressing and frustrating than losing a great vocal performance due to poor recording. A simple "problem-solution" approach to this should have you on your way to capturing great vocals from the beginning. The great news is that the problems are very easy to fix and even avoid completely.

So here's what to look for to avoid bad vocal recordings:

1. Sibilance
2. Plosives
3. Room reflections
4. Proximity effect
5. External noise

1. Sibilance

Problem: Sibilance is a harsh high frequency sound that occurs when humans pronounce consonant syllables (most commonly with S's and T's). Surely you've heard recordings in music or radio when the sibilance is very apparent and not very pleasing. The exaggerated essssssss or ttttsssss can become unbearable quickly and ruin your recordings. Since it is generated from high frequencies, it tends to be even more apparent in female vocals. Sibilance isn't as audible in everyday conversation, but it's there, it's just not as noticeable until you're on a microphone.

Solution: Depending on the microphone and the artist, you're most likely going to get some form of sibilance, which is ok as long as it's not too harsh. Standing a little further from the mic and even off to the side a bit can control the amount of sibilance that occurs.

If you do happen to miss the mark in the recording process and your vocal has more sibilance than you like, there are tools such as 'multiband compressors' and 'de-esser's' that can eliminate the problem as well. This special processing can be done either in the mixing process or the recording process.

2. Plosives

Problem: Plosives ("popping") are the dreaded pops that occur when pronouncing P's and B's into a microphone. These pops are a result of low frequency blast of air that is naturally created when we speak. Much like sibilance, it's not very audible until behind a microphone.

Solution: There was a beautiful invention created to avoid this issue call a pop filter and it does just that, filters out the pops!

There are 2 types of pop filters:

- Mesh filters - usually made of a nylon material
- Metal filters

Both filters work great and will solve your plosive problems. One thing to keep in mind when attaching a pop filter is to make sure that you don't place it too close to the microphone. Plosives occur much stronger the closer you are to the microphone. So depending on the artist, style of recording and microphone, the pop filter may need to be adjusted accordingly.

3. Room Reflections

Problem: The space you're recording in will determine the amount of room reflections you will get. Simply put, the larger the space you are in, the longer it takes for sound to travel creating more room reflections.
Why is this bad?

Well it's not always bad depending on how the reflections in that space affect your recording and if that is the sound you are strategically going for. But if that's not the case, room reflections can turn your recordings into an "echo-y" mess.

Solution: Since most home studios are either going to be in a bedroom, a living room or a basement, you're never going to have a great situation ideal for recording. Those spaces just aren't designed for it. But acoustic treatment such as bass traps, acoustic foam, diffusers and acoustic panels can eliminate a lot of the room reflections and solve the problem. For a more direct solution, **vocal booths and reflection filters** are the way to go.

Also, never underestimate the **power of a closet**, especially if it's acoustically treated. Many people have created great recordings in closets.

4. Proximity Effect

Problem: Common sense would tell us that a microphone is going to pick up sound at different volume levels depending on the distance the vocal source is from the mic. On a technical level, mics also pick up different frequencies in relation to distance as well. Getting too close to the mic results in something known as the proximity effect. The closer you are to a microphone, the more the microphone will pick up lower frequencies. So if a vocal is recorded too close to the mic, proximity effect will cause the vocal to become much more muffled and unclear.

This is due to the nature of cardioid microphones. Because the capsule is placed at the front of the mic, all of the sound passing through it has nowhere else to go but straight in. So getting too close overloads the capsule resulting in a "bunching" of low-end frequencies.

Although proximity effect is most dramatic and intolerable on vocals, it is actually used as a recording technique to get more "low-end" out of certain instruments such as guitar. But this technique requires practice and experience to get it right, otherwise it will ruin the instrument recording as well.

Solution:

- **Pop filters** are a great solution because they won't allow you to get close enough for the proximity effect to occur if they are placed properly.
- **Manage your distance** properly and being aware of what your doing will eliminate the possibility of proximity effect ruining your vocals.
- Using an **omni-directional microphone** will also help decrease proximity effect because the capsule is designed to capture sound from all directions.

5. External Noise

Problem: It's pretty obvious that any noise other than your vocals or instruments should not end up in the recording, unfortunately due to the conditions in so many home studios, it can be very tough to avoid.

There are many factors can contribute to external noise in a recording:

- **Living in the city**: Depending on where you live, noise from outside your place of residence is completely out of your control (sirens, yelling, traffic, trains etc).
- **Roommates and family**: If you live with other people, hopefully you can get them to respect the recording process and if there are kids around, that's a huge noise issue to manage.
- **Moving around when recording**: When recording, it's all about getting in the zone! But for some people, that zone includes snapping fingers, stomping your feet and just a lot of movement in general. Out of control movement often leads to hitting the microphone or other things around you. So keep the energy high but somewhat controlled.
- **Wearing noisy clothing such as nylon**: Now this is a good one that most people never think about but will ruin your recordings quicker than anything on this list. Wearing nylon track suits or noisy material should always be avoided when recording because microphones pick it up like crazy.

Solution: Usually a soundproof recording booth can eliminate a good portion of external noise. If movement is the issue, just be aware of your surroundings and try to channel all of that energy into the microphone, focused excitement is the key.

Conclusion

As you can see, the solutions are very easy and straight-forward for tackling these common recording issues. More than anything, it's about being aware of possible issues and preparing correctly before you begin recording.

Recording With Effects

For many, it's a complete shock when they find out how boring and bland a recording really is without any effects on it. Even a well-performed vocal won't be the most appealing to the ears when it's "dry" ("dry" is an industry term used to describe a vocal or instrument with no effects on it).

Because of the unappealing nature of a dry vocal, many people chose to record with effects to get a little more sense of feel during the recording process. This is completely subjective to the people involved in the recording process and is not the right or wrong approach. In fact, many top engineers believe that you should do as much as humanly possible to get everything to sound right during the recording process so there is less to do during the mixing process. Which is an excellent strategy, but generally only effective with people who have lots of experience recording and who understand the implications of what they're doing.

When new to recording, the easiest and safest way to take care of a dry vocal is to add reverb. Reverb will instantly give the vocal a sense of space and depth. But generally only singers like recording with reverb. Most rappers do not like recording with reverb unless they are performing a melodic rap.

If you plan to use autotune (or some type of pitch correction software), many people believe it should be applied during the recording process so that the artist can hear how the tuning will affect their vocal and they can adjust their melodies, harmonies and pitch accordingly. But if an artist isn't comfortable recording with the sound of their voice being tuned, pitch correction can also be done during the mix process. The issue that comes with recording with pitch correction software is latency. Unless you have a really high powered computer or an audio interface with built in dsp recording processors, you will always get latency when recording with pitch correction software. This can make it extremely difficult to get a proper recording.

Using effects digitally with plugins or externally through analog gear...

In the digital world, recording with effects isn't much of an issue, because if you don't like how the effect sounds when it's time to mix, you can adjust it or get rid of it all together if needs be. If you record an effect through an external analog unit, once it's recorded, it's very hard to get rid of it if you decide you don't like it. We recommend using plugins until you have some more experience under your belt and are more confident in how you want your recordings to sound.

Things to keep in mind moving forward...

- **Take the recording process seriously!** There is nothing worse than having to re-record a great performance because you weren't paying enough attention or had too much to drink and weren't coherent enough to notice mistakes.
- **Do whatever you can to either use a recording booth or to create a makeshift one in the space you're using**. Your recordings will come out much tighter and you won't have to worry about external noise.
- If you live close enough to a music store/retailer, **rent different mics** and experiment with them to see which one(s) you think suits your voice best. It doesn't cost much to rent and it helps narrow down the options before you purchase.
- **If you know somebody who has experience recording, learn from them as much as possible** and ask their opinions on what some good mics are for your budget. It will save you tons of time.
- One thing that is taught early on in recording is - **garbage in, garbage out**. Meaning, it's extremely difficult to make a bad singer or a bad recording sound good. Even with today's great software, experienced mixers can work wonders but you can only do so much.

- **Don't blame your gear on bad recordings**. It's a matter of learning how to get the most out of the gear you have. The "blaming gear" mentality will only keep you broke, **it won't change your results**!
- **Make sure there is absolutely no "clipping" when you record**. Always watch your meters and make sure it's not going into the red. Especially when using digital audio. There is not as much headroom in the digital world and when distortion occurs it can sound very nasty and can't be fixed.

8. Recording Instruments

There's no doubt that learning to record instruments can be tricky for beginners, but it doesn't have to be. For so many of us, we want our music to sound like our favorite records. And because of this desire, we develop unrealistic expectations of how our music should sound in the beginning stages.

Just by toning down your expectations you can make the recording process much easier and enjoyable. Even though you might know how to play an instrument(s) really well, you need to understand that you don't know how to record them well. Exercise some patience and give yourself time to figure out how to record your instruments properly and effectively.

Here are some basic fundamental techniques to get you on the right track…

Recording Guitars

Electric Guitars

Connect the guitar with a ¼ TS cable to the amplifier and mic the electric guitar amp cabinet closest to the grill facing the center of the driver with a dynamic microphone.

They should be recorded hot (high level) under 0db. Move the dynamic mic around for optimal sound quality and judge sound by ear.

Acoustic Guitars

Recording an acoustic guitar can be done with one or two condenser microphones. With one the microphone is typically 9 inches away from the guitar to capture a balanced frequency range aimed towards the sound hole. Move around for desired frequency response.

For recording with two mics, typically one near the hole of the guitar to capture the lower end frequencies at about 6 inches away and one mic near the 12th fret about 6 inches away to get the nice plucks and higher frequencies. Record at just under 0db to get the most out of your dynamic range and the lowest s/n (signal to noise) ratio.

Effects Pedals

Many people who play instruments prefer to use DI (direct input) effects pedals when applying their effects. But using effects pedals is all up to the individual and what sound they are going for. There are a ton of pedals on the market and they all have their own unique flavors and sounds. In our opinion pedals tend to be more useful in live settings so that the musician can adjust the effect themselves if needs be but they can be just as effective in the studio as well.

Like recording with any external effects unit, you have to be careful and know what sound you are going for because once it's recorded it's extremely difficult to change after the fact.

Keep in mind plugin companies emulate many of the popular pedals if you wish to go the software route and save some money.

Recording Drums

Now there are a million things to consider when recording drums and a litany of different ways to record drums as far as microphone setup is concerned. And even though I have recorded many drum sessions in my day, I can honestly say that recording drums is not my speciality by any means. But even with that being said, I want to keep things short and sweet here with just the very basics. But if you do plan on recording drums in your studio, you will definitely need to do your homework in this department. Or you can always just use an electronic drum kit, which is actually a very good alternative these days as the sample packs are incredibly accurate and realistic. But for now, let's do a quick rundown.

Cymbals

Place two condenser mics above the cymbals facing down about 3-6 inches away from the cymbals (closer for less room sound and more high frequency response, and farther for more room sound and less high frequency response).

Kicks

Two Dynamic microphones are usually the best way to mic a kick drum. A microphone inserted inside the drum hole and one microphone about 3 inches away from the kick drum. Move mic around for desired frequency response.

Hi Hats

Hi-hats are typically captured by the overhead microphones but it can be mic'd separately in a similar fashion as the overhead microphone. The only problem with this is that it usually brings a lot of bleed from the snare.

Snares

Snares are typically recorded with two dynamic mics, one about two inches from the top of the snare away from the drummers sticks, and underneath the snare in the center of the snare about two inches away.

Toms

Toms are typically mic'd the same way the snare drum top

Single mic drum recording

If a dedicated drum kit is not in the budget or it is just too much too fast to try to mic a full drum kit, not to worry. All you need is one condenser microphone to capture the room sound of the drum set. With this technique it is common to have the mic centered in the room facing the drum set. From there, adjust the placement of the mic for desired room sound to capture the energy of the room.

When recording drums there is always going to be a certain amount of **bleed** (when the mic picks up the sound of surrounding instruments that are in close proximity to the dedicated instrument being recorded) that will be present in the recording. This is an inherent aspect of recording drums that can usually be managed by adding a gate to the signal. Another way to reduce bleed from other drum sections is to put acoustic foam on the mic stands near the microphone. Also wrapping the kick drum in a couple pillows will help tighten the sound of a kick drum can be an effective technique.

Electronic Drum Kits

Like I had mentioned earlier, electronic drum kits have become an excellent alternative to acoustic drum kits in home studios. In my humble opinion as someone who cannot play the drums at all, I do believe there is more pros than cons to recording with electronic drum kits

Pros

- They are much quieter
- They are much more versatile in terms of the different styles of drum kits
- They can be inexpensive
- They require much less setup
- There are acoustic designs that feel very close to an acoustic drum kit
- They require much less editing
- The samples available are just as realistic as an acoustic drum kit
- They are easier to move around

The only cons to using electronic drum kits is that they don't feel quite like the real thing. But more and more drummers I know are championing electronic drum kits for all of the reasons I have just discussed. And as technology gets better, so does the realism in the feel. And even though these realist kits are much more expensive than the standard ones, they are available. I currently use an electronic drum kit in my studio and it works fantastic!

Recording Keyboards

Keyboards are typically recorded with a ¼ inch TS connection but can be TRS as well. When recording keyboards, it's important to have the keyboard master fader all the way up at 0db to receive the hottest signal before clipping. If the keyboard has onboard speakers you may record with a microphone as well. If you do want to try this technique, it should be done with a mic close to speakers about an inch away, but it is always an important rule to go by ear.

Keyboards should always be recorded to the hottest level below 0db before clipping to retain their full dynamic range with the lowest signal to noise ratio. Keyboards can also be connected via midi cable and can be used to trigger midi data with virtual instruments as well.

Bass

Bass guitars can have two recordings simultaneously via DI box and microphone by the amp (if there is one present). If you choose to record through a DI box, record it hot under 0db with a 1/4 inch TS connection. Next, mic the amp cabinet about an inch away from the grill facing the middle of the driver with a dynamic mic and this should also be recorded hot under 0db. Like all other instrument miking, move the mic around for optimal sound quality and adjust by ear.

Both of these techniques can also be done as their own separate recording method, it's all up to how you wish to do it. Many engineers like to record using both techniques simultaneously because they both give off different frequency responses that can be blended together for a more unique sound.

Conclusion

While there are many technical aspects to recording instruments, try not to get too caught up with it. You wouldn't believe the way some of the instruments were recorded in some of your favorite records. Creativity goes a long way in the recording process. Remember that music is art and should be treated as such in all aspects of the process. Always be open to trying new techniques on top of the traditional technical ways of doing things.

9. Recording Midi

What is Midi?

First of all, midi (musical instrument digital interface) is not a piece of hardware. When some people think of midi, they think of a midi controller of some sort.

Midi is actually a protocol that allows electronic instruments to communicate with digital software.

Most computer sound cards support the transfer of midi data, which means you should be able to use most midi controller devices with most computers. Midi controllers can be connected via usb, midi ports, thunderbolt and firewire.

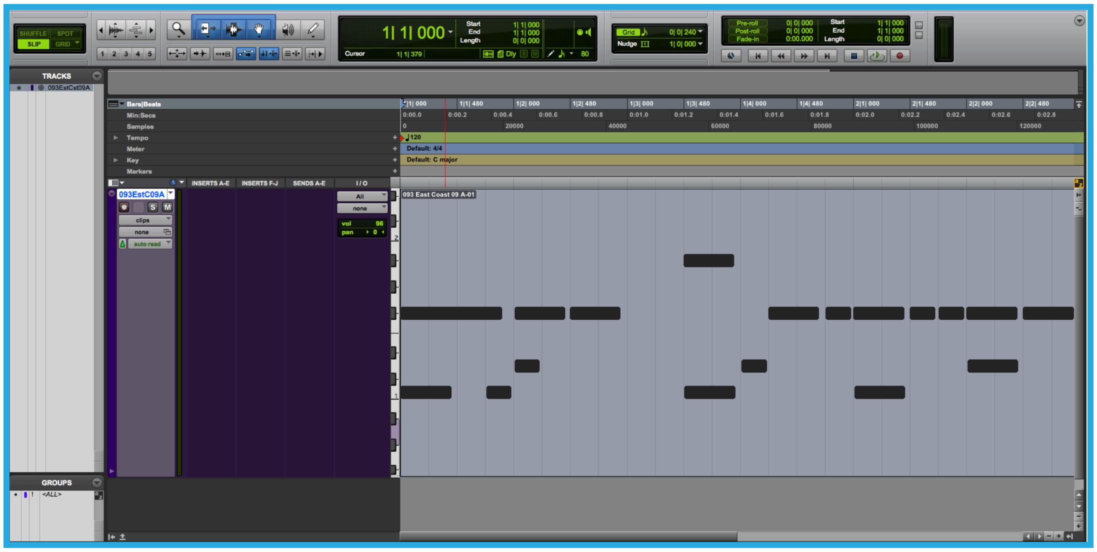
Since midi does not produce sound directly, it uses specific commands to transmit sound information to interpret how music is produced. With today's technology, midi controllers have so much functionality that they can control most parameters on a DAW without the need for a mouse.

The standard functions that midi performs are:

- **Notes** (note on and note off) – Controls when a note is being triggered or played and when it is being released

- **Velocity** (key pressure) – Controls how hard a key or pad is pressed before it bottoms out
- **Pitch wheel change** – controls the signal of the pitch wheel that is pretty much standard on most keyboards.

Recording midi



With most of the newer midi controller devices, getting set up is quite simple. Once the midi controller software driver has been downloaded to your computer, new technology allows the controller to map many of the functions to your DAW as soon as it's connected. While midi mapping is still readily used today so you have the ability to really personalize your workflow with your specific midi controller, if you do want to configure your controller manually, you will need to refer to the instruction manual of your controller to see how to configure it to your specific DAW.

Once the midi controller is connected, you then must load up virtual instruments in a virtual synthesizer to find sounds to be triggered by the controller.
Once you're ready to record, set a metronome or click track and the tempo and go nuts!

Midi notes

If you are familiar with recording audio, you will know that it's recorded as an audio waveform into your software. Midi is typically not recorded to your software as audio waveforms, it's recorded as midi notes. Midi notes are blocks of data that can easily be manipulated by software in many different ways so that you can alter them and move them seamlessly throughout the tracks and musical grid.

Although the basics of recording midi are standard across all recording software, keep in mind that different DAWS have different features. A common problem that producers face with recording digital instruments is that the nature of digital recording requires notes to be played in exact time with the tempo. Even though this can have its benefits, the problem is that it makes the music sound unnatural with no swing or rhythm. On the flipside, it's nearly impossible to play melodies or drums perfectly in sync with a tempo. That's where quantization comes into play.

What is quantization?

The simple definition as it applies to music is; the process of transforming musical notes. What quantization allows you to do is play your off beat notes and align them to the tempo automatically (this can be done during or after the recording). So when you playback your take, it has been accurately aligned to the tempo grid. Many people who have less experience playing instruments tend to quantize their recordings and then move them slightly off the grid after the recording is complete to give it a more natural flow.

Conclusion

All in all, midi controllers have almost become standard in recording studios because of their versatility and the explosion of virtual instrument software. Every sound you can think of can be at your fingertips with the click of a button. As for all of you synthesizer people, you know that your synthesizer can also be used to trigger midi data as well as play the stock sounds that you've come to love, so no need to compromise between the two.

10. Solo Recording

Many of you setting up home studios are in it for yourselves, which means you will most likely be recording yourself, which is fine. If you're doing your recording in a room with no isolation booth, then most likely you will be able to do your recording close enough to your computer to control your recording equipment at the same time.

If your setup involves you recording away from your recording equipment then other measures must be taken.

There are a couple of ways to go about tackling this:

1. The best way to accomplish this these days is with **DAW controller apps**. So if you have a smartphone or a tablet, there are apps available that allow you to control your DAW from your device. Keep in mind these apps mostly

cater to the more popular DAWS on the market, but more and more companies are following suit.

2. If you have a **secondary monitor screen and a wireless mouse and keyboard**, this will allow you to record yourself at a distance as well. Simply connect your monitor to your computer with a cable long enough to reach your recording area and mirror your screen to your primary screen. Now, with a wireless mouse and keyboard handy, you can control your DAW straight from your secondary screen. A stand or stool may be of use to you here as well.

- **Both of these methods will most likely require a headphone extension cable**

Conclusion

To save yourself some grief, think ahead and be sure to set up your room in a manner that makes recording yourself easy and effortless. But don't get too comfortable recording yourself all of the time. The longer you get used to making music on your own, the harder it is to work with other people. Remember that collaboration is what music is all about!