Mastering The Basics

- **INTRODUCTION**

What is Mastering?

Audio mastering is the final stage in the music production process. Mastering is a way of enhancing the sound of the recorded mix by opening/polishing the top end and tightening and defining the low end, without overly changing the sound.

Audio mastering is a process used by audio engineers to transition a completed piece of music from its final mix to the format that a finished product will be distributed from. This involves adjusting the level, equalisation (EQ), and dynamics in various forms such as limiting, compression and de-essing. Mastering can also remove unwanted noise from the mix such as hiss, rumble, or clicks and pops if needed.

However, due to some of these processes audio mastering can be so much more than simply cleaning up a recording. It can bring warmth, punch, and a clarity that a mix engineer is unable to produce when they are more concerned with the individual sounds and balance between the instruments, rather than the whole stereo image.

I would compare mastering to French polishing a table. The table has already been constructed and the parts put in place, all that is left is the finishing process (mastering). If the grain and construction is good, a fine sand and clear lacquer will do. If not, then a thick coat of paint will help cover the cracks.

After the sound of the mastered tracks has been approved, a mastering engineer will take all mastered tracks, edit the fronts and ends make any creative fades, and then balance the levels between the tracks. Adding these final changes transform the individual tracks into a single album/single. Finally, the mastering engineer will change them into the format needed for distribution and duplication.

You may not believe that since you don’t intend to sell your music you don’t need to take this final step and get your music mastered. However, without mastering your music, when played next to other mastered music it can sound unfinished, lower in level, and have an amateur feel.

Using high-end professional mastering engineers can sometimes be expensive so you may want to master your mix yourself. Personally I would not advise this because it’s always a good idea to have some fresh ears make sure the balance and final audio of your mix sounds as good as it can. Since you have been listening to the mixes over and over, your ears will be adjusted to hearing an unfinished sound. However there is plenty of software and information available for you to master at home. So regardless of budget, if you're serious about your production, mastering is always a good idea.

Mastering : A Brief History

In the early days of music production, mastering was limited to preparing the final mix for copying. Engineers often started their recording careers in the mastering studio with hopes of moving up to recording or mixing for artists and the mastering process was left to the newbies. It was not considered a creative part of music production and was thought of only as a way to gain good ear training and practice for would-be engineers.

At this point in history, mastering was necessary so that the frequencies of a mix were altered to fit the requirements of vinyl records. Before copies of a record could be produced, the music needed to be mastered to ensure that there would be no distortion when the record was played on a standard record player. Without mastering, vinyl records would be unable to produce all the sounds of the recording. In extreme cases, the distortion on the record would be so bad that the needle would actually jump across the surface of the vinyl.

In the early stages of recording history (prior to 1960's), the studios were owned entirely by record labels, but when independent labels started cropping up in the late 1960's, the role of mastering began to change. People began to realize that mastering was capable of altering the sound of a recording in other ways. It was still being used only to ensure the playability of a vinyl record. However, mastering could be used to improve the quality of sound as well. Any frequencies / dynamic issues that might occur in the recording studio could be fixed and the final sound of the recording could be perfected.

Artists and music producers wanted to be able to make final changes to their albums after the mixing and the space for the mastering engineer to take part in the creative process was born. Mastering engineers became responsible for making the changes after musicians left the studio and added the extra polishing producers were looking for.

Attended and Online Sessions

Modern music is almost entirely recorded in a digital format. There is no longer the need for a razor blade in the studio and the idea of rolling back a tape is unheard of. That is not to say that no recording studios still use these methods, but by and large everyone is using computers. Computers have changed the name of the game in almost every industry under the sun. The recording industry is no different. Technology has revolutionized the music business in a big way.

One of these revolutions is in the quality of sound. Mastering, as mentioned before, is the final touches, the little fixes, and small changes that makes a couple of tracks into a finished album. Beyond that however, mastering takes recorded music and turns it into a finished product. A finished product that is ready to be heard, bought and sold.

The widespread use of digital formats and computers allows artists to send their music off to be mastered remotely. Internet transfers of files also allows for artists to choose from any mastering studio in the world to finish their album saving budget on travel costs means using a more experienced and therefore more expensive engineer.

Historically, artists would have to attend sessions in person to bring the tapes along, but the internet and computers allow for more freedom in the mastering process. Artists can still meet engineers in person to discuss the sound as the process is taking place, but they can also send their music to a remote mastering studio along with their instructions / reference tracks for how they want the album to be completed.

This can have the advantage of the producer listening in the environment they know best and then giving quality feedback to the mastering engineer to make any tweaks.

From my stand point as the mastering engineer it is nice to meet the artists and get into their head space for the day to completely understand the sound they want to achieve, but mainly due to budget restraints this isn't always possible so a few emails / phones calls and reference tracks usually achieves the same result. It's always cheaper using online services as you are charged by the track rather than the hour with attended, prices are kept low as the engineer can fit online work around their attended sessions (downtime) so with this in mind it's best to not look for price, equipment and the engineers experience in your genre is the key to achieving the best result.

- **EQUIPMENT**

Room Setup

There are nearly infinite possibilities for purchasing equipment to master your tracks and just as many ways of setting up a room to use as your studio. However, the best results come from recognizing how acoustics work and using the acoustics of your room to your advantage. A mastering engineer needs to be able to hear the music properly so that tiny changes can still be heard. This is why it's extremely important for the engineer to know their room sound well, I don't believe there is ever a perfect room but like learning new techniques with the equipment it's important to keep making small adjustments to the room to help improve the sound.

With this in mind it's an important aspect of the room that everything should be the same day to day. Adjusting speakers, moving equipment, or sitting in a different place will change the acoustics of the room and how the music sounds. So you need to do these changes as little as possible so you can test the sound and know why it happened instead of moving everything and not being able to keep track of the rooms acoustic tweaks.

Tips :
These are the very basics that will gain the biggest sound benefits before you get into small changes such as leads,power supplies and buying loads of different equipment :

- You need to position your speakers in a equilateral triangle with you sitting perfectly in one point of the triangle.

- Make sure you have all first reflections points covered with broadband absorbers, not foam or egg boxes :)
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- Always try and place your studio in a rectangular room with you sitting facing the shortest wall.
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- Fill the back wall with as much bass trapping as possible.

Equipment

What computer software will you need? Are you going to work digitally or with analogue equipment? Does the music you are editing need a specific set of tools? What can you do with each piece of equipment? Each of these questions will be answered in turn and the basics of each method and piece of equipment will be explained.

Computers have truly changed the way edits are made to a mix. However, analogue equipment is still used and preferred by most mastering engineers. Which of these will work best for you is determined, in a large part, by budget and skill. There is no point in buying top of the line equipment if you are a beginner, even if you have the money to spend on a professional grade studio. It is always best to start small and gain the skills you need to use the equipment properly. Education and practice are the most important pieces of equipment for any engineer.

This section will lay out the options as far as: What equipment to buy, how to set it up, what the pro's and con's are of each piece, and how to determine what you need.

Source and Destination Software (or 4-point editing)

Source and destination refers to a process of editing where the source audio is preserved on one track and edits are transferred to the destination which is saved on another track or even multiple other tracks. Computers allow for digital copies to be preserved, and edits to be saved to different tracks. Therefore, if you needed to access the original track before any edits are made, the source track will be available to you in its original format. Source and destination software enables the mastering engineer to hear what the edits sound like in comparison to the original recording.

Source/ destination is also capable of chopping up a sample of music. Track One will display an image of the wavelengths for the piece that is going to be edited. By setting up a source track and a destination track, the engineer can choose a destination location at any point on the destination track. When a destination has been set a portion of the audio can be highlighted and transposed to the destination location on the destination track. The highlighted portions determined what part of the source audio will be moved and the destination determines where the source audio will go.

Edits can also be made where the destination is in the middle of other edits. This function is called ripple and it basically moves all other edits over to make room for new edits.

Editing with source/destination is extremely easy. All the engineer needs to do is determine what piece needs to go where. The hardest part of source-destination is figuring out exactly where the start and end points need to be from the source audio and where the best location is for the destination audio to make the edits sound the way they are meant to. Placing and cutting edits relies on the skill and talent of the person making the edits.

Source and destination can be placed on the same track as well. This method is usually used to take pieces from one part of a track and repeat them again later in the same track. For instance, if the engineer wanted to repeat another verse or line from a song, they could use source/destination on the same track to copy the exact same audio to a later point in the song.

The same track method can also be used to place a section of a song much further along in the track. Then edits can be made to fine tune and adjust edits in sections by zooming in on the piece of audio in the destination.

Source/ destination is relatively easy to use and understand, but using it correctly and wisely is the challenge. Every tool is only as effective as the person using it. However, this particular tool is relatively easy to master and can be used to accomplish some very complicated edits.

Source/ destination editing becomes very useful when different parts of an ensemble are recorded separately and/or if multiple takes of a song are recorded. The engineer making edits to the piece can find the very best sections from each instrument track and easily reassemble them into a final track. So if the drummer or guitarist get off beat in a track, but the track sounds perfect otherwise, edits can be made to take the very best parts from each take to make a perfect track.

Additionally, by setting up multiple source and destination tracks, edits could be made in a kind of staging area. This area will allow the engineer to hear the edits before sending them to the final destination. Then when they are satisfied with the edits a final track will be assembled.

This method can also be used to make radio broadcasts or add in clips that were not recorded in the studio. Editing this way can be as simple or complex as the project at hand requires it to be. However, this tool is extremely versatile and useful.

There are a number of software technologies available that will allow users to work with this technique. My own research shows that most recording software, i.e. Cubase, Sonar, Sequoia, etc. have a function that allows for 4-point editing. However, many opinions exist out there and determining which editing software is the best is nearly impossible.

The best software would be easy to use and the number of steps would be minimal to perform each task. They layout and design (what you see on the screen) needs to both be appealing and visually easy to understand. However, the right program depends mostly on the individual using it, what their preferences are, and what skills they have. However, there are many pre-packaged software programs out there that include many different mastering tools in one. These would be ideal for someone who is just beginning to fool around with mastering or someone on a fixed budget.

Analogue Hardware

Some artists prefer the sound of analogue to digital and will prefer that their music be mastered using an analogue system as well. Analogue formats use different equipment than digital and is considered to have a wider range of sound than digital as well. Analogue is often referred to as warmer in comparison to digital.

Digital formats need to translate the natural wavelengths of sound into a language that a computer can understand. This means a great deal of the sound is lost in the transfer from analogue to digital. Some engineers and artists believe that this loss of wavelength results in a flatter, duller sound, and have decided to keep every step of their musical process in the analogue world. This way they lose nothing during production.

Analogue mastering equipment is typically pretty expensive. This is because compressors, equalizers, tape machines etc. all need to have vacuum tubes or other hardware to manipulate the sound frequencies.

Additionally, each piece of equipment usually has only one purpose. So for different effects to be achieved while mastering a piece of audio, multiple piece of hardware will be needed.

Thankfully though, for those engineers out there that prefer to work in a totally analogue system, there are plenty of manufacturers still producing analogue hardware.

Digital recordings are often argued to sound harsher than analogue recordings. In Bob Katz's book, *Mastering Audio*, he claims, "that the accuracy of digital recording reveals the harshness in our sources, since digital recording doesn't compress (mellow out) high frequencies as does low speed analogue tape" (Katz 197). This is one of the reasons that some engineers prefer to work with digital equipment. The argument is claiming that the over accuracy of digital recording captures sound in a way that is overly accurate. Then instead of hearing the audio the way it would sound if it were played live, we hear the harsh reality of what the frequencies are actually telling the microphone. However, Katz argues that with higher quality equipment, the edgy and harsh sounding recordings can be avoided.

Digital is not always better than analogue the same way analogue is not always better than digital. Beyond that some digital is not better than some digital and some analogue is not better than some other analogue. The best way to judge a piece of equipment is to hear the result of using it. Judging for oneself is the best way to chose which equipment is best.

Digital Plug-ins

Digital plug-ins operate the same way analogue equipment does. So if an engineer is using an analogue compressor, the digital equivalent will perform the same task. The only difference being, how the task is completed. Digital plug-ins are computer programs that are designed to reshape and format audio with the same outcome that its analogue equivalent would. The question of effectiveness as far as digital versus analogue is a heated debate and some may disagree that the outcomes are equivalent. However, they perform, what is essentially, the same job.

There are tons of digital plug-ins available on the market that perform a wide variety of tasks. These plug-ins can be digitally installed on a computer and many audio engineering programs will come with at least a handful of them pre-installed.

One large advantage that digital has over analogue, in this instance, is space. Each piece of analogue equipment needs its own space, but digital plug-ins only need enough room to fit on a hard drive. Then once uploaded, an engineer can access all of their plug-ins in the same place and the only limit to how many different plug-ins are available is room on the hard drive.

However, if the choice is between an analogue piece of equipment and a digital plug-in, the best way to decide which one is better is to try and find a way to hear the equipment or the plug-in in action and to find manufactures that have a history of producing quality equipment.

Additionally, digital plug-ins are highly versatile. There is a plug-in for just about everything. If an engineer wanted to make a guitar sound like it was being played through a specific amplifier, there is a perfect plug-in available for him to do so. Or if someone wanted to add an effect to a piece of audio, every effect out there has a digital plug-in version and there are always different versions of the same effect to choose from.

Plug-ins are available to manipulate audio in every conceivable way. There are even plug-ins out there that mimic the sounds of famous amplifiers, effects, and instruments. If an engineer wanted to make something sound like it was being played through a Fender Showman Amp with classic fender reverb, there is a digital plug-in designed to do so.

What kind of plug-in, depends on the software available. Virtual Studio Technology (VST) plug-ins are, by far, the most common and work in both PC's and Mac's. However there is also AU for PC and Direct X for Mac out there as well.

Hybrid System and its Pros/Cons

In a hybrid digital/ analogue system the best and worst of both world is potentially available. However, with decent equipment an engineer could reap the benefits of a dual system. Finding the right equipment is the challenge here. Getting the best equipment available may mean getting only analogue or digital depending on the tastes of the engineer. However, that may not be economically available.

Cost is usually the most limiting factor in purchasing equipment, so the question changes from: What is the best piece of equipment to complete this project, to what is the best piece of equipment available for X amount of money to complete this project?

Analogue to Digital Conversion

Converting audio from an analogue to digital format requires software that is capable of reinterpreting analogue frequencies into a digital language. During this process the analogue signal is quantized into absolutes. This means that the computer, into a specific range of steps, reinterprets the continuous range of frequencies that audible sound consists of. These steps are represented by a digital value. To break this down, think of these steps or intervals as a set of stairs versus a slide. While the slide is continuous and cannot be represented with a number value, each step in the staircase can easily be assigned a number to represent it. The computer can only understand input when it is told exactly where a signal must go. Analogue is not limited in this way.

When converting from analogue to digital and quantization assigns the analogue signal to an absolute value or step, errors do tend to occur. A process called **dither** often accompanies the software used by engineers to convert audio into a digital format**.** Dither will be more fully explained later in this ebook, but put simply dither reduces the negative effects of these errors by extending the range of signals that the A/D converter can interpret.

Monitoring

Monitoring is, possibly, the most important aspect of mastering audio. Monitors are the speakers that play back everything to the engineer. Those speakers need to be of a high enough quality to hear every part of a track or song as clearly as humanly possible. With poor quality or cheap monitors the mastering process will be impossible to complete to satisfaction. Mastering engineers need to have quality monitors to be able to effectively make the microscopic changes and adjustments to the music.

Studio monitors are designed to give a clear and honest interpretation of a recording. They do not add effects or attempt to make the audio sound any better or worse than the recording tells them to. They provide a candid interpretation of what the recording actually sounds like and they do so in a very high quality so that no accidental or unintended interference occurs.

The purpose of studio monitors is not to make a recording sound better than it really is, but to make it so the engineer can make work with the recording to make it sound as close to perfect as possible.

Monitoring requires great attention to detail, a high degree of ear training, and equipment adequate to handle the task of playing back full range audio recordings honestly.

Speaker Selection

Monitors come in passive and actively powered models. This is one of the first choices a buyer would have to make. Passive speakers are powered by an external amplifier, while active speakers have a built in amplifier.

As mastering studios are affectively built to be the ultra hifi room, most mastering engineers will choose a passive pair of full range speakers. This then allows them to listen / match (geek out) different amplifiers and cables for which sounds best to their ears and their room acoustics. Some of the brands that are used heavily by the top studios are Bower & Wilkins, PMC & Dunlavy.

Frequency range is another important aspect when selecting monitors. Most project studios or non commercial studios will use mid range speakers for monitoring, these are great for getting detail in the important frequency range of between 80hz to 14k where most of the musical information is happening but the normal range of human hearing spans from 20 Hz to 20 kHz. This is one of the main reasons mix engineers will use a mastering room as the ultimate listening room to hear super low and high frequencies in a controlled space. Therefore the goal is to find monitors that will produce a clear signal at all frequencies within a conceivable range of sound.

Some monitor speakers with come equipped with intelligently designed settings, such as an acoustic space setting. Acoustic space allows for the user to account for the placement of the speaker. So if a speaker is set against a wall or in a corner, the speaker will adjust to those acoustics. These are best left alone as they are compensating for a badly performing room and will not playback a genuine sound. The same goes for adding graphic equalizers to the amp again you are better to treat the room not the speaker.

Space, as always, is another limiting factor. The size of the speakers may limit some. The speakers need to fit in a certain space so that they can function correctly. For instance, some users may need to place their speakers on a desk or table along side other equipment. Doing so could change the acoustics of the speaker output or produce unwanted distortion from the surface they are set on. To avoid this mounts or stands are preferred and at the very minimum an acoustically absorptive surface to set the speaker on is needed.

Full range

Full range monitors come equipped with tweeter and sub woofer in one package and are able to produce sound across the entire range of human hearing. Lows, mids, and highs are all producer by the same unit. These speakers are typically a two-way design, meaning that sound is produced by both sides of the monitor, front and back.

The advantage here is that these speakers usually sound very good and save space at the same time. There is no need to purchase multiple units to cover the range of human hearing.

Mono/Stereo

Most mixes in modern music are produced using stereo sound. This means that the speakers playing the audio back are producing sound the same way a persons ears pick up sound. Half of a mix is played through one side and the other part is played through the other side. Mono refers to the opposite; both sides are playing exactly the same thing.

However, not everything in the recording is reserved for one side or the other. Oftentimes, recording will produce sound that is hard left (only coming from the left speaker) or hard right (only coming from the right speaker), but also everywhere in between. Most music will use a combination of stereo positions. For instance, the drum track could sound dead center, while the rhythm guitar and bass tracks are leaning one to the left and the other to the right.

Monitors are often sold in pairs on the cheaper end and as single speakers on the more expensive end. However, without a pair, all audio will sound mono, even if they were recorded and mixed in stereo. For this reason, it is recommended that a pair of monitors, at least, should be in every studio so that stereo sound is always an option.

Surround??

Metering

Metering is only as good as the ear that is listening to the sound. Meters provide a guide for the overall volume of a track and can tell an engineer where the track is peaking. However, metering is also one of the most useful tools an engineer has. All tracks have a dynamic range that need to fit into the requirements of an audio system. Simply put, speakers, as innovated and technologically advanced as they have become, have limitations. Part of the mastering process is to ensure that signal does not step out of the boundaries of the equipment. After all, the point of recording music is to get people to listen to it and they need to be able to listen to your track on relatively cheap speakers without it sounding harsh or distorted.

Whether you are mastering in a digital, analogue, or mixed environment, meters are only a guide to what is happening in the music. The ear should be giving the final verdict on whether or not the track sounds right. To get a balanced sound from track to track, the mastering engineer needs to *listen* and listen closely. Tracks recorded at different times or in different places will not meter exactly the same due to the different acoustics. So listening and making sure the tracks sound right is the best way to make sure they are right.

The importance or metering when mastering

Metering is very important when it comes to mastering because it provides a crutch that allows you to ensure that the track being mastered is not clipping. Typically, when clipping occurs (clipping is when the meter bounces into the red) distortion is going to find its way into the track. While distortion can sometimes add new elements to music, unwanted distortion that occurs from clipping pretty much always sounds bad.

The mastering process will often aim to maximize the overall volume of a track. This is why it is important to pay careful attention to your meters to make sure that the added volume is not adding unwanted distortion. As always, the meter is only an aid, the final verdict should be coming from your ears. If it sounds like something is wrong, there probably is. If the track sounds good, then there is probably nothing wrong with it. Trust what you hear and listen repeatedly. However, it is possible to tire your ears, so make sure to take breaks and always play back the track under the same conditions.

What are your options -- Phase, VU/ PPM, Spectrum, and what do they do?

VU Meters or volume unit is a basic meter design. These meters portray an approximate value of RMS (Root Mean Square) or the average voltage level. This put simply, means that the meter tells the engineer what the average level of signal is for a sound.

Although these meters have been around since the beginning of recorded music and are still being used today, they are relatively cheap meters. They do not fair well when used to measure the levels on complex sounds and generally read lower that the true signal strength. Short percussive sounds are picked up differently than long sustained sounds because VU meters display the average level of the signal. Even if the snare hits and horn blasts on a recording reach the same peak, the VU meter will not give them the same values.

VU meters are useful for ensuring signals are getting through to a particular channel, but do not tell you where the peaks are on a track. Since VU's do not have a peak sensor, they are much cheaper than just about every other alternative.

VU meters are still widely used in the audio recording industry and are the favorites of many engineers. VU's help to visualize the character of the track and to portray where the music is going and coming from.

PPM *(peak program meters)* are much more valuable and useful for mastering. Even though the engineer should be making all final decisions with his or her ears, PPM's tell where the track is peaking and aid in making decisions about limiting or compressing the sound. However, these meters are often referred to as "Quasi-peak meters" because they do not register many subtle peaks on a track. PPM's also will typically incorporate a slow return from peaks to aid in visualizing where peaks occur. This helps the engineer *see* where the peaks are happening.

Some meters will use the best of both quasi-peak meters and VU meters. They will typically have a bar or dot above the VU display that lingers at each peak before it falls off.

RMS

Phase Meters are used when mastering a track in stereo that needs to be compatible on a mono system. In some cases, signals that are the opposite of each other will cancel each other out when switched to mono. This means that parts or the entire track will be lost when switching between mono and stereo. Phase meters help to indicate whether the track is going to be capable of being used in a mono environment. The scales used for phase meters usually have a +1 or 0° on the one end and -1 or 180° on the opposite end. Signals in the negative half of this range will usually result in problems when converted to mono. This is very important if you're planning on producing vinyl because if the bass is out of phase the stylus will pop out of the groove.

Spectrum Meters measure the various signals on a track, over the entire frequency range. This means every single frequency from the lowest lows to the highest highs on the track can all be viewed in one place. The peaks are visible for all parts of the track and the frequency of the peak is visible. This means that you can tell when the bass is peaking separate from when the vocals or guitar are peaking. Additionally, when unwanted distortion occurs from not enough headroom on a particular frequency, the engineer can easily see and adjust to fix the track.

Competing with the Volumes of Today

Everyone wants their music to be as loud as possible and they want this for a very simple reason: loud music sounds really good. However, mastering is not only about trying to get the most volume out of a track. It is more important to make sure the sound comes out balanced. The main goal of mastering is to make the track sound its best. Besides volume, mastering needs to give clarity and depth to your tracks. Mastering will always result in a larger and louder sound, so it is not important to worry too much about volume. Gaining clarity and giving tracks depth, greatly improves the sound of the track and volumes are boosted at the same time.

Mastering sessions balance overall volumes of tracks to create an overall volume level for all the tracks. Generally, everything is going to get louder to get this balance. However, new equipment and technology is allowing mastering engineers to really push the limits of their volumes. Music is getting louder and louder and a lot of that volume is coming out of the mastering studio.

Analogue or Digital meters

Just about every kind of meter available today can be purchased in analogue or digital formats. Many engineers and musicians will explain at length why one is always preferred over the other. However, choosing a meter should really be based on the budget and requirements of each individual. In addition, the meter should really be considered by the virtues of that meter. The best way to choose is to purchase the best equipment that you can afford.

Many engineers will defend analogue over digital formats vehemently, but digital is very quickly become a much better contender. Obviously some digital meters will be better than analogue meters. All this means is that some meters are better than other meters, regardless of digital vs. analogue.

However, analogue or digital, meters are only guides to explain what the music is doing. They do not know or understand how the human ear works or perceives sound and they cannot be trusted completely. Mastering must always come down to the human ear and careful listening. Meters are helpful to hint at what the music is doing, but they should only guide the engineer.

What to watch

What are we looking for here? I have been watching a ton of YouTube stuff. I’ve also found some other useful books at a nearby bookstore. I could provide some links in this section to some particularly helpful videos.

- **TECHNIQUES**

Getting ready for mastering

When the recording is finished and the mix is complete it is highly important to set the mix up so that the engineer who is going to master your music is able to complete the production effectively. Levels should be adjusted so that they are not bouncing into the red and should hover around the -5 mark on the mix bus output. If there is not enough room left the master will end up cutting off the peaks. A good mastering engineer will be using professional compressors and limiters that will round those peaks off, rather than cut them off which can add distortion to the track. Digitally crushing the mix so that those peaks are squared, rather than rounded off will add unwanted sound and distortion that will essentially destroy the sound of the track.

The bottom line is, leave enough room for the mastering engineer to do their job, but feel free to add any compressors or effects to the mix you please, but avoid over compressing / limiting your mix. Everyone wants your final product to sound its absolute best and crushing the sound down with digital compression that squares rather than rounds the peaks will not help your track to sound its best.

Is the Track finished?

Even if your final mix sounds excellent and you think there is nothing that could be done to make it any better, mastering is still important. The mastering engineer will be able to go over the track with a fine-toothed comb and fix any problems that are not easily noticeable.

Getting music mastered in a studio is never worthless. A good mastering engineer will be able to listen to your tracks with highly trained ears in an environment that they know extremely well. They will be able to make your tracks sound their absolute best and give balance and added volume that is crucial to professional grade music.

A great deal of modern music is recorded in home studios that lack the professional grade equipment and environment that makes music sound its best. By getting your tracks mastered, especially demo tracks; you are giving your music a competitive edge that will prove to the music world that you are serious about your music. 99.9% of the time, no track is finished until it has been mastered.

Is the mix exactly how you want it?

Making sure the final mix is exactly how you want it to sound will help the mastering engineer enhance and improve the quality of the recording. If the levels are off or the mix is not complete, the mastering engineer will end up performing the job of a mixing engineer rather than mastering the tracks. The ME does not want to do this and you should not want this either. Making sure the final mix is exactly the way you want it to sound lets the ME do their job, not yours.

Additionally, for online sessions it is important to give the mastering engineer some reference material for the sound you are going for. This cuts down on back and forth time by letting the master know exactly what you are looking for. Find some cd's or mp3's that have a sound you want to hear out of your own tracks and they will have a much better idea on how to proceed.

Stereo buss Processing/Limiting

The stereo buss is used to process multiple channels or tracks on one single stereo channel. For instance, when recording multiple instruments or using multiple microphones to record one instrument, it is much easier to add effects or adjust levels on a single stereo track. This also consumes a lot less CPU memory since you will only be working with one channel rather than many.

A new stereo buss is needed for each step in the recording process. Generally, this will happen twice; once when working on a final mix and then again when the music is ready to be mastered. The master buss is what is going to be mastered.

At this point limiting and compression is going to be used to correct any problems that are on the track. Compression should be used as lightly as possible. Compression shrinks the dynamic range of the music, so using too much will take away the natural punch of the track. Limiting needs a light touch as well. Mastering is about preserving the sound of the music, while improving the clarity and quality of the sound. Using too much of either compressors or limiters will make the track sound flat and uninteresting.

Submission criteria

EQ

Equalization (EQ) is used to adjust frequencies across the entire spectrum of sound in a track. Mastering engineers may use parametric or graphic equalizers to fix frequency problems. EQ can be used in Bass, Mid, and Treble range to remove unwanted sounds, such as distortion.

Microphones pick up sound in a fundamentally different way than the human ear, so equalization is very important to bring a balanced sound to the mastered track. Adjusting EQ is also needed to eliminate feedback (that hissing or buzzing sound a speaker may make) in a mix.

Monitoring is very important in this part of the mastering process. In order to achieve a good frequency balance in the track, high quality studio monitors are needed to accurately portray the sound. Monitors that have a full frequency response.

Mid Side

The mid-side EQ is used to separate the stereo signals for the mastering process into “mid and “side”. In a typical music mix, all bass instruments, including drums will be placed in the mid channels, while the panned element elements of the mix will be placed on the side channels. As a master engineer, the mid side separation will provide you with more flexibility when it comes to giving either the mid or side channel a high-end boost. Mid side EQ will also allow you to easily get rid anti-correlated signal that is often present in low bass range. It can be quite a radical way to eq so a cautious approach is needed it can ruin somebody’s hard worked mix quite quickly.

Filtering

Among the many techniques applied by master engineers in their work is filtering. This can be done with the specific circuitry of an amplifier which will only allow certain sound frequencies to pass during the processing of audio.The range at which it operates at is between 0 Hz to 20 Hz. An audio filter used in the equalization gadget can either cut, pass, or enhance the sound frequencies it was calibrated to affect.

During mastering, engineers apply either low or high pass filters to the audio they are processing. Low pass filters are filters that cuts off sound that have a frequency below the limit set and conversely high pass filters will cut off sound that are at a frequency higher than the limit set. Using their experience, they are able to reduce the noise that can be heard on the mix with these tools.

However useful, filtering can introduce some issues to the mix. It is due to the nature of the filter to cut out any and all frequency that don’t pass the set requirement. This can be troublesome when dealing with different sounds within a mix since it requires more of a surgical approach due to its complexity.

Staging EQ from piece to piece (Analogue/digital)

The staging process of your mix when you master audio can depend on the process that you apply as well as the equipment that you may have. The choices are between analog or digital processing.

Both produce a uniqueness in the processed audio which would create a lot of different options for the master engineers to choose from. This can cause some sort of confusion because certain audio will sound better when you hear them but when it is measured on a hard scale measurements are too far off from each other.

Thus, comes the need for balance and knowing what to do and what to apply when it come to choosing how to process the audio mix that you are held responsible for.

If you choose a digital process, you should first check the limitations and weaknesses of the devices that you are using. With that knowledge, calibrating the digital system just right will be easier.

There are many things to consider when you start to stage your mix. First off you have to set a certain bit limit to your sound passes. Following that would be the audio mix/es you're handling, conversion quality, the equalization of the sound, the mixes in general, and the gadget used during the processing.

When choosing between the two, engineers can actually apply the good from both types to cover up their own weakness. The important things is to find the balance that can help produce the best music.

Terminology

Compression

What is compression?

The main function of compression is gain reduction. This is a process that hugely involves regulation of the signal amplitude in order to achieve a uniform level of sounds within the mix. Compressors are used balance between the peaks and dynamics of the music mix. They are considered as reinforcement tools and are the next popular tools to equalizers. A good compression process will ultimately result to a punchier audio.

Gain reduction can be achieved by manipulating the ratio and threshold controls. The ratio is used to control how much grain reduction you will apply based on the pre-set threshold limit/setting. Gain reduction for signal peaks should ideally be between 3dB to 6dB. This is one way to ensure a stable signal without creating too much pump in the entire mix.

Mid Side

Mid side processing can be further done to an audio mix that sounds too narrow. Using compression have the tendency to affect the width of the sound, you can address this by doing a mid/side processing. Core parts of the mix such as the vocal , snare, and bass should remain in the mid/center though, while you can distribute other elements of the mix to the side. You can use any of the trusted M/S processing tools to ensure you do this accurately, such as Ozone and Brainworx.

Another way to ensure that you avoid narrowing the width too much during compression, is to compress the mid more, as you increase the volume on the sides at the same time. Using mid/side in your mix will result to a wider top end, while also creating a tightened effect on the bottom end .

Multiband

Hailed as one of the kings of mastering techniques, multiband compression is one of the easiest ways to come up with a flawless audio mix as it provides the master engineer the versatility, which is not that available in full-ban compression.

Note that when using a full band compression,any change you make in the settings will be applicable to the entire frequency. This is the main and difference and advantage with multiband compression, as it allows application of a distinct compression setting for every band. This means one band can have a different ration and threshold, and attack and release time from another band. Multiband compressors are made up of several filters, with distinct settings, which means there are multiple frequency bands in it. For instance you want to compress the low frequency bands such as the drums, you can do this, without having to compromise the other mid-range to high frequencies. Once you have meticulously adjusted each signal, you can recombine these to finalize the mix.

Compression Staging

Compression is a process best applied during the mix. While some people tend to do it in the earlier stages of recording, mainly for their guitar instruments, a more practical and organized way to do this is to wait for the entire mixdown. Consider compression as part of the initial preparations for mastering. An ideal audio mixing process is as follows:

- Equalization
- Compression
- Limiting
- Noise reduction

Similar techniques will still be applied during the mastering process, but compression will be mainly used to adjust the audio width at this stage.

Limiting

With a similar gain reduction function, limiting is often confused with the functions of compression. To get things straight, we establish here that unlike compressors that rely on threshold levels for signal control, limiters have their own "limit" controls, which will eventually influence the audio level passing through it. Normally, compression comes before limiting, as the latter may only be needed for peak-limiting. This applies to both mixing and mastering stages. EQ precedes these two all the time.

Dessing

- **DELIVERY**

What is Dither?

Dither is one of the final steps in the recording process. Dithering is an intentional application of a form of noise to the recording. The process is used to limit quantization error. Dither is simply the process used to make sure quantization error or quantization distortion does not happen in the final recording. Dithering is one of the smallest and simplest steps in the music production process, but can also be incredibly important. Without dithering some music will simply sound awful.

Quantization error occurs in digital audio because of the particular input from the original audio.

In digital media coding and sequencing occur to turn the actual wavelengths recorded in the studio into a language a computer can understand. Even though computer systems are getting more and more complicated, the basic format still exists. Computers only understand absolutes. So when wavelengths do not fit into one of the predetermined steps that the computer understands, quantization occurs.

Quantizing is how a computer decides to fit analogue signals into digital formats. When the computer receives an input it must determine which step each portion of the audio must fit into. Rather than a continuous range of signals that blend from one into the next, computers understand these signals in something like a staircase. Step one is below step two, which is below, step three, and all of the first three steps are below step four. Nothing can be in between steps; they are quantized to which ever step is closest.

This process usually works very well. There are hundreds or even thousands of steps and the human ear is not precise enough to pick up on all the miniscule differences between the continuous sequence and the step-wise sequence. However, sometimes the computer will mess up and what should have been interpreted as one thing will be interpreted as something else, or multiple something else’s.

This is why dithering is so important before a mastering can be complete. This small part of the final product eliminates all the errors that can occur from quantization error.

Dithering occurs when the audio is converted from its original analogue format to a digital format. This is when quantization errors will occur so dithering must be done to immediately correct quantization distortion. However, when some edits are applied to a piece of audio, like compression or equalization, the frequencies are being changed. Because the computer is now reinterpreting where these signals land, dithering may need to be reapplied.

Simply put, dithering is the process used by engineers to fix or avoid quantization errors. This is a small part of the mastering process, but without proper dithering a piece of music could be made unlistenable.

What are Sample Rates?

Ever since the audio industry started recording songs/audio digitally, the use of an Audio-to-Digital converter (ADC) became the standard. The conversion process involves the ADC taking samples of the electric signal produced by the sound being recorded and quantifying that to have a digital value which is the language that can be read by a computer. This device follows a specific periodic interval before it takes another sample until the entire audio line is completed. Its periodical approach, called **sample rate**, is measured in Hertz which has been proven to have the minimum error in capturing the signal from the original sound source.

In the early days of digital audio recording, experts have identified that the minimum sample rate should be at 40 kHz so that they can record sound properly using the new method. Sound engineers adjusted this rate to 44.1 kHz when they started recording on CD’s to compensate for the technical requirements of this media.

However, many audio experts and pundits alike think that this sampling rate has limitations which could affect the integrity of audio being sold to consumers. The popular opinion is that though a 44.1 kHz sample rate is enough to hear the sound there is a possibility that the original sound can’t be captured in its entirety.

The reason behind this is the anti-aliasing filter which does not accept frequencies that is higher than 20 kHz. There are instruments like chimes and certain cymbals that produce sound in higher frequencies which the filter leaves out during recording sessions that uses the 44.1 kHz sample rate.

Though frequencies above or below the 20 kHz range is deemed as “invisible” to the human auditory system, audio producers are sentimental to the point that their whole creation isn’t digitally recreated in full. This is the reasoning behind the suggestion to record at higher sample rates. In doing so, digital conversion will be closer to perfection and filters can be more accepting to sound waves of higher frequencies.

Along with that, many are pushing for the idea because it is already feasible right now when it comes to the budget and the materials that can handle this kind of material. Unlike in the past when producing digital audio in this frequency was expensive, technology has progressed enough to make it a cheap option and has also helped develop the possible mediums to be able to handle audio produced using higher sample rates.

Despite all these advancements, there are still technical issues that need to be addressed in the production of this type of sound. A few experts have said that this method might reduce the quality of the final product.

A lot of questions regarding sample rates are still up in the air and people are just going with what they prefer in terms of audio quality when they are buying the music they want.

Dither-Sample rates
Here's a simple guide to dither sample to help you with your audio engineering :
[photos of sample rate conversion here]

Industry Standards
We are undoubtedly living in the digital domain now, and this greatly affects how and when we use dither. One main consideration here is the fact that most of the CD recorders today can only store up to 16 bits, while the usual audio mix output is 24-bit. This is when we use dither noise to ensure that all the information in the original output are still available in lower bit. In the music industry these days, dithering is regarded as one of the processes that assures numerical accuracy.

Through a high definition compatible disc (HDCD), it is possible to alter gain structure up to 20bits (or even better), since its encode-decode system has been known to have one of the best A/D converters in the market to date. Going from 24-bit to 16-bit dither can be a very huge degradation, which translates to almost -91dBFS. However, an average listener will not be able to notice this given high precision dithering.

Dither in Practice

Here are few more dithering practical notes you can use along the way.
* While the main function of dithering now is lowering of the sample rate, make it a point to convert the sample rate to its shortest wordlength at the end of each operation. You can still perform some immediate dithering along the way, but these should be subtle, somewhere between 48 to 24 bit, for instance.
* If you can avoid it, do not go as long as 16 bit for your records. If this cannot be completely avoided, give HDCD a try, instead.
* Dither have distinct flavors or types which may not be generally applicable for all music types. Make sure to explore these types first and examine whether they are fit for your music.

* Dithering is not equivalent to adding news to the record. Dithering is mainly used the extend digital system resolution, which leads to eliminate any distortion that may have been caused by quantization.

Stereo Image- techniques

Stereo imaging is the major polishing that you can do to your record. Part of the enjoyment of the music is hearing where the sound is actually coming from (left-right, front-back end). Boosting the stereo image will specifically enhance this listening experience.

First trick is to achieve more width in your audio, by for instance adding more mono elements within the mix. When it comes to stereo imaging, it's really all about creating that credible perception of width and audio enhancement. Another common technique among sound engineers is altering the frequency in each side of the image, using either EQ or filter plug-in. Use copies of the audio files, pan each on left and right channels, then apply the intended EQ or filter son both. This stage is basically just about creating that contrast and balance in the audio.

Panning in the mix

Among all of the stereo image techniques, panning is one of the most effective especially in creating that trick where distinct sounds seem to be coming from a specific side of the stereo. In actual application, both sounds are actually present in both channels, however, through channel, a listener is able to distinguish sounds coming from a specific side of the image.

During panning all low-frequency sounds should be kept at the middle, this includes the bass, kick drums, and vocals. Consequently, you can strategically pan all mid to high-range sounds on each side for a more entertaining stereo image. Try panning these sounds between 3-9 oclock position, rather than doing the extreme panning from the center as you may be at risk at having unnecessarily wide stereo image.

(Ken Townsend Abbey road, contribution studio presence Abbey road studios)

Ken Townsend made breakthroughs in stereo imaging through his automatic double tracking (ADT) invention, which modern sound engineers now conveniently use to duplicate signals that can be used on each side of the image. He was the man behind one of the most successful Beatles of all time, the Revolver, where the use of ADT was already incorporated. He likewise played a significant role in establishing one of the world's most renowned recording studios, the Abbey Road Studio - - where the ADT was consequently first used.

Phase (?)

Efficiency

Session Recall

Templates

A Starting Point

Media Formats

With the rapid advancements in technology, there are now more options for audio formats upon mastering The era of analog formats lasted until the late 70s, with vinyl records hitting the markets big time. Simultaneous to the releases of vinyl records, the Betamax digital audio was also introduced in the market. Today, Mp3s are currently the top player in this industry, thus being used by most master engineers, as well. Let's take a look at some of the leading digital media/recording formats today:

- Mp3 – one reason why mp3 files are very famous among master engineers is its ability to retain a good quality of the audio, despite the fact that it compresses to almost 1/10 of its original size. Since people are now more concerned about acquiring and storing files, mp3 files proves to be a good option for storage, especially for music files. This is however not a recommended format for voice files.
- Wav – a format that is specifically designed for Windows computer units, makes use of the uncompressed version of the CD-extracted files. This explains why .wav files have larger size, but the sound quality remains as high as the CD-quality. For lighter file needs, wav files can still be reduced using different codecs such as GSM and MPEG-3. Apple has a similar standard audio file in the form of *aiff .*
- Wma – Windows Media Audio format was developed and owned by Microsoft, which is one of the pioneer audio file formats that provides copy protection capabilities under the Digital Right Management design.
- Aac - Advanced Audio Coding is the file format adopted by the iTunes Music Store for all the copyrighted music that they sell. This format is originally owned by Dolby, which is under their MPEG4 audio standard. Apple did a copy-protected version of it for iTunes.

These formats are among the most used music format mainly because of the number of available software where they can work. For highly specific needs, there are still a number of emerging formats like *dvf, atrac, mid,* and *flac.*

- **TIPS & TRICKS FROM THE PROS**

We end this book by few important points that we have learned and tried from the experts themselves, which you may also find useful in your own mastering endeavors.

- With the overwhelming number of mastering tools today, it can be tempting to use them all at once. This is not how successful mastering works. The trick is not to use these tools all at once, and instead settle for those you are most comfortable with. A great combo for a start would be a good parametric equalizer, compressor, and enhancer.

- Always have a basis for comparison before mastering. This can be a couple of CD tracks that can be your benchmark for your work. This will also help you detect inaccuracies in the monitoring system a lot easier.

- For most starters in mastering, dithering seems like a complicated process to even undertake, but it’s one of the crucial processes needed especially when there is a need to render the audio to as long 16-bit. It ensures consistency in quality, even if the difference may not be that noticeable when you listen to your track after dithering.

- The old cliché “If it’s broke don’t fix it” strangely applies to audio mastering. Do not overdo things, just for the sake of using the tools. Some recordings actually sound better as they are.

- Use crossfades sparingly. Using 10-30ms will already suffice, and doing more than that can create a double-tracked effect.

- EQ and compression should always go together, to maintain the balance in the mix.

- While it is really better for mastering to have good monitoring speakers, but if these are not available, the trick is to listen to the track through different gears from headphones, speakers (big and small), and car stereo among others.

- Don't rely too much on limiters for loudness. If you want to achieve that commercial loudness in your audio, it is the combined job of accurate EQ, a bit of compression ,and limiter. For best results, do not use more than 3dB of gain reduction through your reduction. Expert in the field could even achieve that punch in their music using less than 3dB.

- At times when you are dealing with tracks that have been recorded in either different studios, or different time, use parametric equalizer in order to even them out. Chances are these tracks will not have matched levels, so you'll have to really experiment with the EQ to address this.

- There are times that an analog system is better than a digital one while it can also go the other way around. Even though the control for digital levels might seem to lack quality alone, if it’s a part of a system there’s a chance that it can improve on that and be comparable to the analog. The trick to that is to set it’s attenuation on the lower end of the spectrum.