MASTERING
Part III: Tools of the Trade

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Mastering studios are definitely different than Recording and Mixing studios, and here we'll look at the differences. We're also going to look at some of the popular gear used by Mastering studios and how these processors are different than equipment found in typical recording studios.

The Room

There are mastering facilities that have been built from the ground up for mastering, but there are also others that have treated an existing room for it. It seems that rooms that are shaped like a shoebox are better than square rooms, since the rectangle-shaped rooms have longer lengths between the front and rear walls, allowing more acoustic control. Ceilings and floors play a role also, and most dedicated mastering rooms are 10 feet in height or taller.

The materials used for a mastering studio can be drapes, carpeting, woods that are absorbent that help control sound as well as fiberglass/foam paneling to trap and diffuse frequencies which are placed in corners, walls and ceilings, depending on where they are needed. It's not all about trapping sound; reflecting materials are also used where there are issues where sound waves clash within a room creating "standing waves" so wood flooring and diffusers are applied when needed.

When building mastering rooms, an experienced acoustician may be hired to design a shoebox-shaped room of about 25'L X 15'W X 12'H. There are several types of preferred dimensions for building rooms, typically based on physics theories by Morse, Bolt, Sepmeyer, Louden and others. When building rooms, Acousticians will most likely refer to one of these recommended "room ratios" which suggest a perfect sound environment, and ideally, all mastering rooms should be perfect but most mastering studios around the world are not, so they are treated for the sonic deficiencies that are inherited with each room's design.

The idea is to control the sound reflections bouncing around the room, hyping or canceling out frequencies. The first reflection should be you (and your ears). This is the reason why mastering rooms usually don't have much more than the loudspeakers, amps, treatment and processors, with the mastering engineer seated directly in front of the speakers (distance varies by speakers, room and that room's reflection signature). If you've ever known an audiophile and seen an audiophile's setup, this is much like a mastering suite with regards to the listening environment.

Recording studios have different rooms that use their sonic signatures while tracking instruments. There are recording studios that use different materials, like carpet, wood floors, wall paneling, heavy drapes, marble, etc. to create and control the reverberation of instruments and capture the sound of the room. Vocals are recorded in treated rooms that reflect little or no sound back to the microphones, which is important for projects that want to add vocal effects at mixdown. There are cases where people track inside rooms and capture the room's reflections for an effect, like tiled bathrooms, if that's what the project calls for.

Mixing studios are essentially control rooms, where you have all the tools necessary to shape the mix, such as the console and effects processors. Nearfield monitors are usually on top of the console, and some studios also have soffit-mounted loudspeakers, useful when the producer and artists are there for the mix session; these "mains" are supposed to fill the room with sound, allowing everyone to hear the mix as loud as they want. Diffusers are common in control rooms, since you want the sound to be spread out throughout the room, although the environment is not as neutral as a mastering room, and mix engineers usually rely on more than just the mains to mix, like another set of monitors and even headphones.

So what can you do if you don't have the budget to build a room for mastering? You can treat a room as much as you can, and learn the sonic signature of that room (which sounds nicer than saying "the room's problem areas"). It's not a matter of completely covering the walls in foam either, because proper placement of your speakers is also important, and having an unobstructed path from them to your ears is essential.

You may also want to consider headphones as an additional reference to your speakers. These days there are audiophile-grade headphones and headphone amplifiers that can help you hear an accurate representation of what you're doing to the material, but without the true stereo image perception of a set of high quality loudspeakers. I'm sure you've heard how it's *just not right* to use headphones for mastering material; well, my opinion is that they're a great addition to a set of high-quality speakers. Headphones can let you hear things that your loudspeakers (and room) would probably hide, like small clicks or pops, and in my experience with a set of nicely burned-in Sennheiser HD650's, I can hear ±0.5db changes over most of the frequency spectrum (emphasis on *most*, not all, which is a reason why having a great set of loudspeakers that can let you hear small changes through a wider range of frequencies is best). You're probably curious as to why people will tell you not to use cans for mixing or mastering, and here's probably why:

Headphones aren't "flat". This seems to throw off most who feel that anything that isn't designed to be flat (when referring to frequency response) shouldn't be considered for critical listening. Sound hits your head and the outer parts of your ears before your brain processes it, so by the time sound enters your ear canal, the frequencies have many peaks and dips, which if measured, would look nothing like that flat line across the frequency spectrum listed on those charts that came with your speakers, and this is the concept behind the *Diffuse Field EQ curve* that is applied to most headphones that claim to have a "flat frequency response." Unfortunately, this standard applies to an "average" head shape, so the sound of any pair of headphones with a DF curve can vary from person to person. The only thing that keeps the DF method of measure in check is an actual standard by the IEC (International Electrotechnical Commission, 60268-7:1996, pg. 61). The IEC defines itself as *"The leading global organization that prepares and publishes International Standards for all electrical, electronic and related technologies."* Again, this method is based on what someone considered an "average" size head, so if you have a large dome and big ears, you'll have a harder time using headphones for critical listening.

But back to the room, you don't *have to have* a specially designed room for mastering, but you definitely need to treat the one you plan on mastering in. How to treat your room for mastering? Fortunately, these days there are plenty of manufacturers of acoustic panels that have websites with a ton of useful information to help you control reflections in your space. Do some research online and check out their sound absorption ratings, the quality products have reports that they're proud to share with customers (also make sure the materials are fireproof). Many of these companies offer great advice and you might even be able to send them pictures of your room for them to tell you how much treatment you'll need just by looking at your space.

Another thing you can do to help you figure out how much treatment you need is to take measurements at your listening position with an SPL meter. This usually involves playing various frequency tones, set at a specific volume level as well as noises (white, pink as well as other spectral noise types) and taking db measurements with the meter. You would then use a chart to make notations as you go through the frequency spectrum to get a rough idea of how much, in terms of decibels any given frequency is being picked up at your listening location (by placing the SPL meter where your ears would be with the use of a tripod).

"Ah, I see, so then once I know where my room falls short in the frequency spectrum, all I need is an EQ to cut/boost the frequencies and I've got a tuned room!" you say? Not so fast. By using an EQ to make up for the differences, you can actually make problems worse by augmenting or diminishing other frequencies at your listening position, as well in other areas in your room. I believe that it's best to treat your room with bass traps/diffusers and if there's a problem that just can't be solved due to some kind of structural issue, then an EQ may help, just make sure you use an EQ that will not introduce distortion, phase, or "color" to the sound. Keep this in mind when considering solutions that offer to treat your room with the use of correction software (EQs) alone.

Through the years, I've learned that a lot of the mastering engineers whose work I'm constantly impressed by don't have perfect rooms, a lot of them have learned to adjust their hearing to the *sonic signature* of the rooms they work in, and while that may mean that they're not hearing a 100% accurate representation of the frequency spectrum, they're getting real close and know how to compensate for it, and if it works, meaning, the work they do for their clients who listen to the material over systems these guys are not familiar with, then it means that they're doing their job, and that's what counts at the end of the day.

Speakers, Cables, Amps,

You can easily spend 20 grand (2008 dollars) on a pair of decent high-fidelity loudspeakers, matching amps and cables. Higher-quality speakers do their best to reproduce a flat frequency response for the listener, and in a room that is treated to tame the problems an enclosed space has as much as possible, a high quality set of loudspeakers will give you an accurate representation of the material playing through the system. There are well-known brands and models of speakers that have been proven to work well in mastering studios around the globe; these models are also typically accepted in the audiophile community as well. The size of the room has to be taken into account, as well as the placement of the speakers in the room, and the distance between you and the speakers, so a good way to go about getting the best speakers for you is to do some research as to which speakers will work best in your space in addition to treating the room. If you plan on spending a few grand on loudspeakers and setting up your listening room to make the best use of them, start hanging around audiophile forums on the web, you should also pick up magazines related to high-end audio, these will expose you to many other alternatives for speakers and amps than what you might pick up from mastering engineers who surf web forums, or by checking out their gear lists online (I've noticed most mastering engineers that talk about what speakers they use on the web usually stick to just a few models of high-end speakers, which are considered to be great in the audiophile community, but like many other things, there are high-quality models out there that are as accurate, but are lesser-known). You might even be inclined to build your own speakers, and there are plenty of online and print resources on this as well.

Focal's flagship speakers are not likely to be seen in your usual gear magazines & catalogs. These Grand Eutopia EM speakers cost approximately $270,000 and are almost 7 feet tall. I've had the pleasure of listening to the previous generation of the EMs, the BEs and nothing truly compares to the sound.

Don't be fooled into thinking you need to have the biggest full range speakers matched with the biggest amplifiers either; you should first consider the room where you will be placing the speakers as well as the intended position. I'm not going to call anyone out, but I've seen pictures of Mastering rooms where the speakers are obviously a little too big for the rooms. This can require a lot more treatment than using speakers that are better suited to smaller rooms.

Keep in mind that many of the leading manufacturers of high-end loudspeakers apply similar design concepts to their lesser-priced models; this is a benefit for those that don't want to shell out the dough for the flagship models, since manufacturers haven't had to spend as much on the development of their cheaper models as they did on the research and development for their flagship units.

The whole cable game, if you ask me, is out of control. It's my opinion that a good set of shielded cables for your speaker system is sufficient, but there are people out there that swear that sound is greatly improved by using cables that cost a lot more, which have "features" that most people don't consider essential for speaker wire, such as cables that are oxygen-free. I simply have not been convinced to try a great deal of expensive cables to confirm this, but to me it makes more sense that the components used throughout your system are what dictate the quality of the audio, and not the cables.

From the research I've done and from what I've picked up from a speaker repairman I've dealt with for years, the most important issues with cables are thickness and length. I've been told thick copper wire (10-12 gauge is usually what most audiophiles go for, the thicker 10-gauge for subwoofers mostly) is what to look out for and it can be purchased almost anywhere, including your local hardware store. Keep the length of the cable from being excessive; don't purchase 50ft. cables if you only need 12 feet of wire (but if you get a really good deal and have a pair of wire clippers around and are comfortable making your own connections, go for it!) If you walk into a speaker showroom, the salesperson will most likely point you to the most expensive cable they've got, but if you ask someone who is not in the business of selling you cables, but is in the business of speakers, the idea is to get thick copper cable in just the right lengths.

Amplifiers come in two flavors: solid state and tube. For mastering, solid state is preferred over tube amps by many masterers as tube amps introduce harmonic distortion, which many audiophiles prefer over solid state amps because of the *warmth* they impart on the signal, but this is something that isn't desired for mastering audio, since we don't want to enhance the sound from the source, instead we want the cleanest path possible from the source, through the amp, to the speakers. For this reason, Amplifiers are used instead of Receivers to minimize the path as much as possible. Receivers are used by home audio enthusiasts to connect various consumer devices, such as CD/DVD decks, radio tuners, turntables, tape decks, etc. Mastering engineers require a more sophisticated solution for this, with specialized features that go beyond the basic source switching capabilities of the home audio receiver.

The Mastering Console

The mastering console is essentially the command center for the Mastering Engineer. Some MEs build their own devices, like monitor controllers and patchbays (the build quality of typical patchbays found in recording studios won't cut it for high-end mastering). There are manufacturers who build custom modules and make consoles tailored to fit the needs of the ME. There are also modular units that can be incorporated to fit into an ME's setup, based on their needs using the most passive designs possible.

A lot of thought needs to be placed on how the mastering console should be configured, how many connections are needed for the mastering chain, what features are useful and it's also a good idea to have an outlook on future expansion, while maintaining the cleanest path within the console as much as possible. In the hardware world, passive devices that don't introduce a meaningful amount of noise are complex, and therefore expensive. For engineers that are completely "ITB", stacking plugin effects won't add noise to the signal, and while the trip out of the DAW to high-end analog can be very worthwhile, it can be very expensive if you plan on doing it with high-end passive gear in mind.

The console can be in a desk that resembles a typical mixing console, with all of the processors laid out within arms' length. When it's done this way, there is typically an unobstructed path from about the ME's torso to the speakers, but because the first reflections from the speakers are hitting the lower portion of the desk before the ME's ears, which will affect the sound to a degree, some MEs choose to put their consoles (including processors) in a rack behind them, or in another room altogether, and sit directly in front of the speakers, with no obstructions, and will have control over the audio, with remotes for monitoring and patching, and processors within reach. Those who prefer to go with a desk, however, choose designs that do their best to not interfere with the sound of the speakers; round corners and low profiles help with this issue.

The features found in mastering consoles vary. Some MEs work with stereo material only, some with Stereo and Surround, etc. Here are some key features that can be found on most serious mastering setups:

Monitor Control – High quality controllers feature stepped attenuators (in 1db steps, typically) as opposed to the "smooth" potentiometer-type of volume control. When dealing with volume control, the quality of the "tracking" in an attenuator relates to how well and closely-matched the two channels (L/R) stay with each other in the range of attenuation. Potentiometer controls don't track as well as stepped attenuators that use rotary, stepped switches, making these less prone to phase and balancing issues when going up and down the audio taper (or volume "curve"). Having multiple sets of monitor inputs and outputs and being able to mono the source, dim (to user-set levels), polarity flip one channel (to check a mix for phase issues) and mute (to answer a phone call or to check to see if that ringing is in your ears only, or in the mix) are also highly desired options.

Patching/Routing – A lot of mastering engineers stick to using a set chain of processors, some always EQ before compressing, and might have for example 2-3 hardware EQs and 2 compressors and will simply bypass the units they don't need (truly passive units pass signal even when the units are powered off), but there are other MEs who might like to switch the order of the processors and/or audition one of the processors in the chain with the press of one button, as opposed to having to bypass all the other processors in the chain, and that's what a high-end router does. For digital devices, there are digital units that route various types of digital connections and formats and some also do sample rate conversion on the fly. This is useful, for example, when you have digital gear that might use AES/EBU connections but want to send the signal to another digital unit that uses S/PDIF.

Mid/Side Processing – Many times, it's not possible to go back to the mixdown stage to correct issues on a mix, and for example you might have a mix where the vocal has too much sibilance. Patching in a de-esser on the mix will also affect the hats, cymbals and any other frequencies in that range. Mid/Side processing (also known as Vertical/Lateral processing in vinyl cutting, and Sum/Difference in other processors) will take a stereo track and combine the information that is shared by the left and right channel (Mid) and separate the remaining difference (Side). What you end up with, after decoding the signals, is the ability to treat the Mid information independent of the Side information of the mix. Assuming that the vocals are panned mostly in the center, you can then insert a de-esser on the Mid channel and not affect the Side channels. Alot of times you can introduce phase by changing the balance of the Mid and Side channels, and affect the stereo image of the mix. This can, however, help in widening a mix that sounds too "narrow". The ability to independently process the Mid information and the Side information can many times save a poorly-balanced and thin-sounding mix.

Visual Tools

A good set of meters is essential to anybody, but there are other visual tools that are very useful to mastering engineers when processing audio. Mastering engineers with loads of experience can hear most problems in mixes without relying on visual tools, but when it comes to analyzing audio, these tools help define problems more precisely and aid when making corrections for things like Stereo Imaging and Phase. In addition to high-quality meters, some tools you might see in a mastering studio are Spectrum Analyzers, Correlation Meters and Bit Meters.

Signal Processing

Contrary to the *"don't do it, you need a million dollars-worth of outboard gear!"* campaign that you might stumble upon on the internet, the processors used by Mastering Engineers are often times plugins these days (and while you *can* spend thousands on them, plugins tend to cost a lot less than their high-end analog counterparts). A decade or so ago, plugin designers had to cut corners due to the lack of high CPU power in computers, and therefore plugin effects weren't serious. It all boils down to the fact that in order to make effects that sound great, they require quite a bit of CPU processing power and it just wasn't feasible to design a plugin compressor that would cripple your Pentium II computer with one or two instances. These days, however, designers can take advantage of more powerful computer processors and can model hardware processors in plugin form that closely resemble the characteristics of the original hardware, as well as create new effects that do things that would be really hard to accomplish in hardware, like a linear phase multiband compressor/expander/gate, for example.

Then there are times when some nice analog harmonic distortion is in order, which happens often in the world of digitally composed music and this is one of the areas where high-end outboard processors outshine plugins. Some high-end mastering EQs and Compressors have a way of treating the signal with transparency, and yet a "depth" that just hasn't been accomplished by plugins. Often this difference is more apparent with mixes that leave a good deal of headroom for the masterer who might have a few nice high-end outboard processors (another good reason to leave more headroom in a mix for mastering). As far as features, plugins can do as much and sometimes more in terms of processing than hardware effects, but in many cases, when a sense of depth and warmth is desired, hardware processors simply sound more pleasing. I like both, so rather than focus on why one is better than the other, which will always vary from project to project, I'll stick to the *types* of signal processors used in Mastering, which realistically are in use in both software and hardware form by mastering professionals out there. Let's start with one of the most important pieces of the chain, assuming your mix resides in binary code somewhere on a hard drive, and that it has to first be converted from zeros and ones to become an electrical signal that will drive a speaker:

DA Converter

Before you even think about a monitoring system that will give you an accurate representation of your audio, you should first look at the quality of your DA (Digital-to-Analog) converter. A great DA converter will accurately translate your data into a signal. Designers of cheaper, more-bang-for-your-buck converters will put lesser-quality components in their units (like many other higher quality devices, the best are sophisticated and cost quite a bit in research dollars and skilled engineering manpower hours to make). Many times, "pro-sumer" converters sound decent, but when you compare the sound of these to a great set of converters, you might be amazed at what you'd hear. What sounds "good" on a set of decent converters will sound amazing over better ones; a lot of times you hear elements and a spaciousness in your audio that you couldn't quite make out before. The audio has a greater depth and it's not because the better converters enhance the sound, it's because less than great conversion doesn't translate the data as accurately as the better converters do; really important if you want to hear your audio as accurately as possible.

AD Converter

Whenever I hear people say that recording digitally sounds harsh, thin, lifeless, etc. I assume these people have never used a high-quality Analog-to-Digital converter. The cheap ones *can* sound thin and lifeless, but better converters will give you the ability to record your music better than anything else that's been available in the past, and when I say "better", I mean more dynamic range, with a smaller signal-to-noise ratio than ever before. In mastering, AD converters are typically stereo converters since they are used at the very last step of an analog chain before hitting the DAW; stereo mastering at the time of this writing is still the most popular delivery format. In the world of high-end AD conversion, there's no question that they're able to capture every detail that's coming into the analog stage of the converter, so the variances in the quality of conversion can be referred to as "flavors". The flavor spectrum of high-end conversion can go from very clean and transparent to some converters that are capable of imparting "analog character" to the signal, and it's not unusual for mastering engineers to invest in more than one of them.

Restoration Tools

One of the first tasks a mastering engineer often tackles is the clean up of noise issues on mixes before applying any other type of signal processes. This can include hiss from tape media, clicks or pops caused by badly clocked digital devices, DC offset & RF noise introduced to the recordings (which happens often when sampling from various sources), and any other type of unwanted elements. In this department, the most sophisticated tools are software packages that feature a comprehensive set of tools. While these tools can work wonders, there is usually a tradeoff, and many times you can't completely get rid of the unwanted noise without also removing some of the material you want to keep; the more serious the noise issues are, the harder it is to completely remove them from the material.

Equalization

Equalizers designed for mastering have features that cater to working on program material. For example, many times they'll have specific frequencies they work with that have proven to work best for mastering, as opposed to having sweepable frequency controls. They also have notched controls for highpass/lowpass filters, gain and Q (center frequency) settings. These controls help a Mastering Engineer quickly recall settings should they need to go back to a track to make changes, and having notched controls for the Q settings also helps with phase compensation between the Left and Right channels, since having a very narrow Q setting on the Left channel, for example can cause smearing of frequencies if the Right channel has a wider Q setting.

Hardware EQs, due to the fact that they are analog (nonlinear) in design, will always introduce some amount of *phase distortion*, and many designs claim to have *minimum phase*. In the world of digital, however, designers have been able to accomplish what is known as Linear Time Invariant (LTI) designs, in which phase increases linearly with frequency; this is known as *linear phase*. Don't confuse this and think that all plugin EQs are *linear phase*; a lot of them can introduce distortion, as well other artifacts that don't sound pleasing, due to poor design.

It is often suggested at the mastering stage, that if a mix requires a heavy amount of EQ, then a trip back to the mix is probably in order, but many times this is not possible. A surgical EQ, one that can offer very narrow Q settings, different types of filters (slopes) and a wide selection of frequency bands can save the day.

Compression

Notched steps are also preferred in hardware compressors for Mastering, as well as having a wide range of settings, from low ratios (2:1 and higher) to very fast and slow release settings. Mid/Side processing and unlinking capabilities are also desired features, and as explained earlier, being able to independently compress the mid and side channels is something that a mastering engineer might need to do from time to time. Another highly-desired feature for mastering is having a high pass side chain, which can minimize pumping during compression, caused by the compressor acting on the lower frequencies in a mix; with the side chain, you would tell the compressor to ignore a certain range of lower frequencies when compressing.

Some software compressor designs can do things that would be very hard to recreate in a hardware compressor, due to the amount of electrical components that would be required to recreate some of the complexity found in digital processors while trying to retain the quality of the material, such as multiband compressors. These tools, while they are sophisticated, are many times used only when it's not possible to go back to the mix to fix issues. Let's say we have a mix that has a kick drum that is a bit louder than everything and it's not possible to go back to the mix session for whatever reason; if we apply a 2-band (stereo) compressor, we will be processing the entire frequency. You could set the threshold to act only on the loudest levels of the mix, which would probably be the kicks that are out of control, but keep in mind that high frequency material also has high peaks that isn't as noticeable to the ear as low frequency material, so more than likely, you'll also be compressing some of your hats & cymbals. Multiband compressors split the frequency range into several bands using filter banks, and in this example, you would be able to compress only those frequencies where the kicks are at (and everything else in the mix that shares these frequencies).

Enhancers

There are times when a mix calls for more complex processing and treatment is required at the transient level. Sound Enhancer effects typically fall in one of four categories:

Exciters – Add harmonic elements and massage the high frequency bands.
Maximizers – Increase the average (RMS) levels in a mix.
Bass Enhancers – Add harmonic elements and massage the low frequency bands.
Transient Shapers – Apply gain to transients and change the perception of them.

The intent of these processors is to enhance the perception of mixes that might sound "thin" or "small". At the mastering level, these effects are used sparingly, as too much can introduce phase issues or give an unrealistic feel to the material.

Limiting

A limiter by definition is a compressor with a high ratio setting, usually 20:1 or higher. Unlike a compressor, limiters don't have Attack settings because they are designed to have 100% attack, as the purpose of a limiter is to prevent peaks from going over the set Threshold (some limiters use an Input control). Limiters accomplish this by using a "look-ahead window", which means they'll delay the signal to the compressor by a small amount of time, typically less than 2 milliseconds to make sure they grab all the peaks in the material. While typical compressors have a make-up gain control to increase the output level of the material after compression, limiters have an output control that does not exceed 0db, therefore the only change in output can be down from the 0db setting. The threshold or input control increases the signal level to the compressor, causing gain reduction. Average (RMS) levels in the material are then increased as the dynamic range is decreased. Some limiters feature *Enhancing* processes (see Enhancers above), while others are designed to be transparent.

Dithering

Last but not least in our signal processing category is Dither. When truncating audio data from 24-bit audio to 16-bit audio (to be able to play your music back on a CD), the result (loss of data) is referred to as *quantization error*. When played back, a truncated data file is in effect, non-linear, and will therefore have some amount of audible distortion. The process of Dithering introduces a low-level random noise to the signal, and the distortion then becomes more "analog" to our ears. There are different types of dithering algorithms such as Pow-r, IDR, UV22 and you might already be familiar with them because all DAWs come bundled with one type or another. Dithering is typically done at the end of the signal processing chain, when bit reduction usually happens and often times, Mastering engineers will audition various types of algorithms to hear which one sounds best for the material.

In part IV, we'll have a closer look into preparing a CD for replication/duplication. To a Mastering Engineer, burning CDs is no joke! We do some things at this stage to make sure that when you spend your hard-earned money on getting CDs pressed up professionally, they won't be duds.