Modern Mixing Strategies For The DIY Musician
INTRODUCTION

Since the dawn of digital recording it has become increasingly affordable for musicians to build excellent quality home recording studios without having to break the bank. In the years' past, if one intended to end up with a professional quality audio recording they would have to have access to thousands, if not hundreds of thousands of dollars worth of high end audio equipment. This, fortunately is no longer the case. Increases in computer processing speed and power along with decreases in the costs associated with such things have allowed us as musicians to have access to extremely high-quality audio recording equipment at a modest price.

Nowadays, if you have a reasonably modern computer at your disposal, some musical talent and some good ideas, a couple hundred dollars is all it really takes to begin recording great quality tracks in the comfort of your own home. Even the cheap stuff nowadays sounds fantastic in the grand scheme of things. You can go out and spend 200 dollars on a small audio interface that comes bundled with some recording software, spend another hundred or two on a nice microphone (and a preamp to plug it into), another 100 or so on some nice headphones (or a couple hundred for some nice studio monitors), and for well under a thousand dollars you have the equivalent of what would have cost hundred of thousands not that long ago.

This, to me, is a wonderful thing.

Now, if it's true as I mentioned that the affordable gear is indeed of a very high quality, why is it then that we hear so much mediocre, low-quality sounding music proliferate the internet today? Well, one has to remember that the tools are only part of the job. The skill of the one who wields the tools is a much more elusive thing. There's a great deal of technique involved in capturing great sounds and a great deal of technique involved in knowing what to do with those sounds once captured to create a compelling musical presentation. The latter is what we will be dealing with in this book.

In fact, it is my opinion that capturing a great sound is far easier these days than creating a great mix of the sounds that you capture. Now, this is debatable and some might argue that there's an equal amount of skill required for both tasks. And maybe you are right. But I have found out in my experience that most people are fairly comfortable putting a microphone in front of their instrument, putting their headphones on, moving the mic around until they get a sound that they like and then hitting the big red record button and being quite happy with the results that they get. So, why is it then that by the time all of the tracks have been recorded and all of the instruments have been put away, you're still left with a muddy and unfocused “bedroom recording” as opposed to the sounds we're used to hearing on our favorite records?

The answer is simple, most people have no idea how to mix.

Mixing is what separates the men from the boys in this home studio environment. The goal of this book is to provide you with a proven, systematic method that will act as a launching point in your journey to your mastery at the art and science of modern record mixing. To begin our journey, I feel that it's important that we define what mixing is and at what point during the music production process does the mix begin.
SEGMENTING YOUR WORKFLOW

The music production process consists of a few definite stages. The first stage can be thought of as the song writing process. This is where you take an idea “from the ether” and through your own creative powers turn it into a musical arrangement. What I mean by this is creating the harmonic structure. The melodic phrasing, the rhythmic character and the lyrical content of your song. However, for the purposes of this book we are going to assume that you’re already able to create great songs.

The second phase in the production process is often referred to as preproduction. This is the stage where you begin to assemble your musical ideas and create your arrangements. This is also the stage where you begin to plan out the recording process. How many guitar parts are we going to use in this section? How many vocal overdubs or harmonies are we going to use in this section? During the preproduction phase you might even find yourself experimenting with microphones, experimenting with guitar tones, or experimenting with different keyboard sounds. What we’re trying to achieve in the preproduction phase is to organize and collect all the techniques that we’re eventually going to use in the recording phase (which of course follows preproduction).

During the recording phase we determine what microphones are best used to capture the sounds in the way we want to hear them. We decide upon microphone placement, we capture all the musical ideas that we’ve assembled during the preproduction phase and we record them into our DAW or sequencer.

These 3 initial stages of the music production process are beyond the scope of this book. Once the recording has been completed (and when I say completed I mean all the musical events have been captured into your recording environment all the microphones and instruments have now been put away and what we’re left with is a multitrack session composed of our musical ideas which together presents the arrangement of our song) you might think that this is the point at which this mix begins... but you would be mistaken.

Before you dive into your mix its very important that you first set the scene. Get all the elements of your mix to an ideal state and create a clear and defined segmentation between the stages of your production process. All too often I see producers flying through their recording and mixing as they go. A mistake.
For example, let say you've recorded a vocal. You feel the vocal sounds fairly good but it sounds a little bassy. So you pull out an EQ. You begin to treat the bottom end of that recorded track. Then you record a guitar and you discover the guitar sounds decent but it doesn't quiet have enough sparkle on the top end. So, you pull out an EQ and add a little high-end frequency to the guitar track. Does this sound familiar?

Well, what I’ve found is that this “mix-as-you-go” approach often ends up with you mixing in circles and never actually arriving at the final polished result. The problem with mixing as-you-go is that your mix decisions (and there are thousands in any given arrangement) have a cumulative effect. In other words, each decision affects the other decisions that have been made. When you start to make decisions before the entire arrangement has been fleshed out, those decisions may be inappropriate once there are other elements impacting the mix. So, again my advice to you would be to draw a clear line in the sand between the phases of the production process.

Once the recording has been completed and you've put the microphone and instruments away, the first step towards setting the stage for your mix is to ensure that every element in your arrangement exist as a high-bit depth audio file. This means that if you've used software instruments during the creation of your arrangement you're going to want to convert those instruments from their original midi into high-bit depth audio. Also, if you've used any amp modeling on your guitar parts or any other such things be sure that you export those and print them as audio files and import them back into your session.

After you've converted all of your software instruments and amp-modeled tracks into high-bit depth audio files and imported them back into your session be sure to mute the original tracks, backup the presets that you've used and disengage the plugins. This process accomplishes two things. First, by disengaging all of your instruments and amp modeler plugins you are able to free up a great deal of system resources on your computer. As you may have already discovered, VST instruments and amp modeling plugins are very CPU hungry. We will need all of this processing power and ram memory available to us so that we don't ever get hung up on a creative mix decision by a system that can't keep up with our demands.

The second thing that this process achieves is it forces you to commit to the sounds that you've used in the recording phase. If you’ve used a particular guitar amp model on your guitar track printed as an audio file make the commitment that this is the guitar sound that you’re going for. After all if these were the "olden days" you would have put a microphone in front of an amp, you would have recorded that sound to tape and you would have been committed by default to the result that you’d achieved. This is a bit of a psychological hack that I have found to be very effective.

So, at this point in the process what we have is a screenful of high-bit depth audio files ready to be mixed. There are no plugins running. There are no VST instruments running. All of those have been backup and disengaged and now we enter the next phase: setting our stage.
I'm a huge believer in the power of an organized and fast workflow. As much as mixing is a technical process, it is also very much a creative endeavour. One of the worst things that can happen is to be flowing with your own creativity only to be impeded by a technical difficulty. Not only is it frustrating but it causes you to lose your momentum and to potentially dilute the potency of your creative decisions from that point forward. This brings us to the next phase of our mix process which is to organize our screen and our audio elements in such a way that we are able to fly through our mix at the speed of our own creativity.

The first thing to do is to organize your screen. Start off by applying meaningful labels to each track in your mix. For example, if a track that you've recorded was labeled “guitar 2” or “John's guitar part” relabel that to something that makes more musical sense. For example if “John's guitar part” was an acoustic guitar rhythm track then label the track “acoustic rhythm” or something similarly suitable. If there's an organ part that only shows up for a few bars in the bridge you might consider relabeling that track as “bridge organ”. The goal here is to be able to quickly and easily recognize what tracks you're dealing with at a glance.

The next thing that I recommend that you do (if your DAW supports it) is to colour code similar tracks. For example if you have 10 tracks that comprise the drum kit, colour code them all the same colour. Myself, I prefer blue for my drums. I'll take all of my guitar parts and colour code them as red. I'll take all of my keyboard parts and make them purple. My bass part will be orange etc, etc. Again this is done in an effort to be able to very quickly recognize what aspect of the mix you are currently working on.

As a Cubase user I take this one step further and I create folder tracks for each of the colours. Folder tracks are simply an organizational tool. They have nothing to do with signal routing or signal flow. I'm simply able to place all of my blue drum tracks into a folder called drums so that I can collapse it out of the way when I'm not working on it. Not every DAW supports this but if you have something similar at your disposal I recommend that you use it.

Next setup your routing. At this stage we want to create a hierarchy of editing control. We achieve this through the use of group channels or buses.

Let's take drums for example. If I had 10 tracks that comprise my drum mix, I might take all of those tracks and route them to a stereo bus labeled “drum kit”. This way I have access to the entire kit on a single fader in my DAW.
You might take this a step further. For example if your drum mix contains 3 tom tracks, you might create a bus called toms, route the 3 tom microphones to the toms bus which then gets routed to the “drum kit” bus. This way we have a hierarchical level of control. If I need to adjust the high tom I can reach for its fader and process it in whatever way necessary. If however, the entire tom group needs to be tweaked I have a fader that represents all 3 tom mics. If the entire drum kit needs a little tender loving care, I have a single fader that represents the entire drum kit as well. Continue to do this for all the remaining elements of your mix.

For example if you have electric guitar parts and acoustic guitar parts you might group them individually. So, I have a stereo group channel for my electrics and a stereo group channel for my acoustics and then subsequently, a stereo group channel for all guitars. You will see rather quickly as you start to dig into the mix process itself how this effort upfront can dramatically increase the speed of your workflow.

The next thing I like to do when setting the stage for my mix process is to create some standard effects channels. I will need a long tail reverb, a short tail reverb and a delay of some kind. As the mix progresses you may find that you need more effects channels than these basic 3. However, in the very beginning just set up these 3 sends so that they are readily available to throw tracks towards whenever the inspiration strikes you.

At this point in the process you should now have a screen that contains nothing but high-bit audio files that have been labeled for ease of recognition and colour coded for the same purpose. You've also set up your sub-busses and basic routing template so all “like tracks” can be found on a single fader. You've got 2 or 3 reverb delays set and ready to go and we're just about ready to begin our mix.

I urge you not to underestimate the power of an organized workflow to aid you in your efforts at mix mastery.
Gain staging, in a nutshell, is the process of adjusting all of the volume levels at all of the stages in your signal path to avoid clipping at any point during the mix process. In digital audio the zero mark on your meters is the absolute ceiling that your levels are able to achieve before creating a clip. Digital clipping must be avoided at all cost as it is ugly sounding and there's nothing musically useful about it. With this in mind it's important that we setup our levels right from the beginning to avoid clipping at any point in the signal path.

The way to do this is quite simple. Open your DAWs preferences and find the area where you're able to adjust the performance of the meters on your mix console. In every DAW that I have ever used, there is an option to “hold the peaks”. This means as the levels bounce up and down, there will be an upper limit that gets reached and a little line will appear on the meter to indicate this peak. This line indicates the highest level that the track achieved during the playback of the mix.

So, simply activate peak hold metering and let the arrangement play from front to back all the way through. While it's playing take this opportunity to listen through the tracks to see if you can hear any anomalies or anything that needs to be edited out. Once the track has finish playing from beginning to end you will have an indicator on the meter of each channel to what the highest peak was during the performance.

Once you've determined the maximum peak of each track apply a trim plugin (if your mix console does not have a one built in) and reduce the gain of each track and until the peak equals -12DB. For example, if your vocal track achieved its highest peak at -2DB then adjust the trim of that track by -10DB to achieve an overall peak level of -12DB. If your guitar track peaked at +3, (in other words it clipped) then your going to want to reduce the gain of that track by -15 so that the maximum peak that track creates does not exceed -12DB.

If you're using a trim plugin to achieve this, be sure that it is the absolute first thing in the signal path before any other plugin that you may subsequently use during the mix process. Once you've done this for each track in your mix you should already start to hear a degree of balance between the elements of your mix.

You now know for sure that you have lots of headroom to avoid clipping, not just the clipping of the output bus but clipping of any plugins that you might happen to use during the mix process. Be sure to save a new version of your session once this work has been completed because now we begin mixing.
Before we move any further, I think it’s important that we pause for a moment and define the term mixing.

What is mixing? What makes a great mix? These are all important questions to begin our journey towards mastery. You may believe you already know what mixing is and you may be right however, I had been practicing for years before I truly realized what I was actually trying to accomplish.

What we are striving to do as mix engineers is simply this:

“To evenly distribute the audio elements of our arrangements amongst the 3 spatial dimensions that exist between the speakers.”

I’ll say it again: the goal of any mix session, in a basic sense, is to evenly distribute the audio elements found in our arrangement amongst the 3 spatial dimensions that exist between the speakers.

So, what are these special dimensions and how do we manipulate them? The 3 dimensions that we work with in a mix are as follows:

First we have the horizontal dimension. This dimension is dealt with by basic panning and stereo enhancement effects.
Next, we have the vertical dimension which is represented by the frequency content present in any given element.

Finally, we have the depth dimension. Depth represents the perceived distance or closeness of a given audio element to the listener and is typically dealt with using reverbs and delays.

In the chapters to follow, I’d like to dig quite a bit deeper into these 3 dimensions and how to use the mixer’s tools to create the psychoacoustic illusion of 3 dimensional space despite the fact that we only have 2 speakers.
CHAPTER 6

THE HORIZONTAL DIMENSION

The horizontal dimension is probably the easiest to understand. It simply involves moving and placing your audio elements between your speakers from left to right typically using your pan pots (or the digital equivalent thereof). It’s something that I’m sure even a novice engineer has already began experimenting with. The technique is simply to adjust the pan pots of each track to distribute the audio elements from left to right between the speakers. The objective here (referring back to our definition of mixing is) is to evenly distribute the audio elements of your arrangement between the left and right extremes of the horizontal dimension.

More often than not, what we’re trying to achieve is balance. For example, if you have many elements in your mix that are similar in tone and timber, it might be wise to pan them separate from one another. For example, 2 or 3 guitar tracks that have a similar tone to them might sound better or might be better presented if they did not share the same space along the horizontal dimension.

Elements that share the similar tonal characteristics also share the same space tend to compete with one another and cause frequency masking. In such cases, the general best practice is to separate them by panning them to opposite directions.
Something to consider when planning your panorama strategy is that elements that have an abundance of low-end frequency character to them typically fair the best when they are pan directly down the center. The reason for this is that bass frequencies are very high energy and cause the cones of your speaker drivers to move further than high frequency content. Your speakers will perform best when those high-energy low-frequencies cause the speakers to drive in sync with one another. This is the reason you'll notice that in almost every modern musical recording the kick drum and bass guitar are almost always paned dead center. Also, from a psychoacoustic perspective if you pan your kick drum or base guitar off to one side your mix will quickly sound and feel very lopsided.

**The sacred center**

The center of your horizontal plane should be reserved for the most important elements of your mix. Typically the items that you'll find in your in the center of your panorama are the features such as the vocal or solo as well as the driving rhythmic elements such as the snare drum, kick drum and bass guitar.

Of course, the fact that mixing is a creative endeavor means that this is not an inalterable rule. However, it is, in my experience, the general best practice. I prefer to keep only the most important melodic and rhythmic elements in the sacred center of my panorama. All other elements of my mix will be distributed to the left and to the right of those key ingredients.

With larger mixes you may find it difficult to keep the center clear and distinct. You may find that the key elements of your mix begin to seem crowded by some of the surrounding elements. This is an example where stereo enhancement tools may come in handy.

Often I'll use a stereo enhancement plugin on a group bus (for example my guitars or my keyboards or perhaps my drums) to further widen their perceived stereo image. This has the effect of seemingly spreading their left and right extremes outside the bounds of where the speaker physically reside. It's an interesting psychoacoustic effect that uses phase manipulation and delays to achieve the illusion that the sound source is in fact wider than the speakers themselves. This can be extremely useful for clearing the center and creating the impression that there is more space for your most important element to live in the centre of the mix.
The vertical dimension in your mix is represented by the frequency content of your audio elements. This is often the dimension that gets overlooked by novice mixers. So, how does a track's frequency create the illusion of its placement along the vertical dimension. Well, I'm glad you asked!

Consider this: If you were to hear a very high pitch sound like a whistle or a screech, you’d tend to perceived that sound as hitting your body in the middle of your head or right up between your ears. Conversely, if you were to consider a low-frequency tone like a kick drum, you’d tend to perceive that sound as hitting your body at a much lower level. That deep kick drum will hit you in the gut, that high pitch whiny screech will hit you in the ears. This is how we perceive the vertical dimension in a mix.

Just as with our horizontal dimension, it's important that we strategize the best approach for our vertical dimension. If you have too many elements that share the same area of the frequency spectrum they will tend to “mask” one another out and create an indistinct rumble as oppose to clearly defined and audible musical elements. For example, if you have an organ part and a guitar part that are very closely related frequency wise, it might be in your best interest to grab an EQ and adjust 1 of those 2 tracks (if not both of them) so that you create some separation between them on the vertical scale.

To illustrate, you might roll off a little low-end on one element while rolling off a little bit of the upper mids on another which may just create a bit of a “hole” or a space in each instrument for the other one to nestle inside. This will allow the listener to perceive both instruments with an equal degree of clarity.

Remember, our goal is to evenly distribute the audio elements between the 3 spatial dimensions. We want to avoid conflict and masking wherever we can so that the listener can easily perceive all the elements of our arrangement.

It's much easier to hear problems along the vertical dimension when the tracks have yet not been panned. For this reason I prefer to start with this part of the process when I begin my mix. If 2 elements do tend to compete with one another frequency wise, it's often more difficult to hear that competition when they are spread left and right away from each other.

Never EQ in solo

If you solo a track so that you can hear more clearly what your EQ doing, you will almost inevitably be creating frequency conflict that you will be unaware of until you bring the track out of solo and into context with the rest of the puzzle. Remember at this stage, you’re not using your EQ to make a track sound good using your EQ to make a track fit. A subtle, yet important distinction.
The third and final spatial dimension in our mix is the depth dimension. This is where we create the illusion that elements are near to or far from the listener. “How do we do this?” You might ask. To answer this lets discuss for a moment how the human auditory system perceives distance.

In a very basic sense, the mind can determine how far away a sound is by comparing the volume of the direct sound source to the volume and timing of the reflections from the space that the source emanates inside of. This is also how we determine how large a room is.

In a mix, it’s important that we consider both things. For example, if a sound source is quiet far from you in a very large space, the direct sound is going to lose energy the further it travels. Hence, by the time it reaches your ear it would be lower in volume than if you had been standing directly beside the sound source. Also, the sound, as it leaves the source will spread in all directions and reflect off all the surfaces that make up the exterior limits of the space that the sound emanates in. These reflections will then bounce back towards you (the listener) and also lose energy in the same fashion by the time they reach your ear. The mind (being an amazingly powerful tool) is able to, in a split second calculate the difference of both time and amplitude, between the direct sound source and the reflections from the surfaces of the walls in the space that the sound emanates within.

Another thing to consider is the fact that low frequencies have a much higher energy output than high frequencies. This means that they tend to travel considerably further without losing their energy compared to their high-frequency counterparts. This means that the further away a sound source is, the less likely it is that you will perceive the high-frequencies of that sound source by the time it reaches your ear. It’s for this very reason that when you’re out in the ocean and you hear a fog horn, its always a very low bass heavy tone. The goal of a fog horn is to stretch the greatest distance possible to be heard by sailors at sea. Low-frequencies will cover this distance whereas high-frequencies will not. So, the further away a sound source is, the less high-frequency content will exist by the time reaches your ear.

So, how do we use these concepts in our mix. Well, let’s take an example. If you wanted a background vocal track to appear more distant than the lead vocal. What you might consider doing is sending that vocal to a fairly long reverb, turn its level down a little bit because of course, it would loose energy by the time it reached the listener. Also, roll off a little bit of the high-frequency content from that vocal. This will create the psychoacoustic illusion that that track is further away from the listener than something with opposing characteristics.
If you follow these workflow suggestions and continue to hone your skills at placing elements in a 3 dimensional sound stage, you will begin to hear the best mixes you’ve ever heard coming from your own studio! I Promise!

If you truly want to start sounding professional and you want to do it quickly, join me and my community of students over at MixLessons.com. There you will find an in-depth, methodical video training series to take you as deep down the mixing rabbit hole as I am capable. You’ll also get access to an extremely supportive community of students who are making amazing records and sharing the things they’ve been learning.

This book is just the beginning of your mastery of the art of modern mixing and I’d be honoured if you’d allow me to take you the rest of the way. Sign up for the 24 week deep dive today at www.MixLessons.com

Thanks for reading and I’ll see you there!

^Dezz Asante
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