



THE ART OF AUDIO MIXING

MODERN MIXING SECRETS

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INTRODUCTION

This document is an essential guide to audio mixing. It is not a magic bullet, but it should help you understand the fundamentals of mixing music using any music production software.

This guide doesn't cover things like programming, music theory, sound design, production, mastering, or recording.

Mixing will never fix a bad recording; if you put garbage in, you'll surely get garbage out. It's that simple.

For this guide to work well, make sure your sounds are well recorded, and if they are digitally programmed, make sure you have some knowledge about sound design.

All the information in this guide is verified and fact-checked. I'm a qualified music producer and have run a couple of blogs about music production in the past 6 years.

I've also helped a lot of beginners in forums, online groups, and discussion boards.

You know you're getting good information from someone who spends the entire day and year in the studio because I don't have a job; all I do is make music.

I know there's a lot of info online about this subject, but it is scattered across disorganized articles, blogs, forums, and video sites.

This is where you'll get all the important information in one place.

If you're reading this, you should at least have heard of equalization, compression, reverb, and delay. You must also know your DAW very well.

This is an essential guide, and the knowledge can be applied to any reputable music production software.

Please find the glossary for this guide at the end. The glossary has been added to help you understand audio engineering terms or vocabulary.

If you have any questions or comments, please contact me via email:

support@audiospectra.net

MODERN MIXING SECRETS

Welcome to the first module of modern mixing secrets.

In this module, I'd like to share some information with you about mixing in music creation.

The most important thing is to get the basics right.

I will try to explain everything as simply as possible in plain, simple English.

Let's dive right in then.

What is mixing?

Mixing is the stage where an audio engineer gets to polish a song to make it sound as good as possible and get it ready for the mastering engineer to put in the final touches.

In the music creation process, audio mixing involves taking all the individual sounds (multitracks) from a sound recording with multiple channels and combining them into a single master channel.

It is the process of balancing individual recorded tracks to work well together as one.

You can look at it as a car. For instance, a car has a lot of parts, such as the wheels, body, engine, gears, seats, etc., but when those parts are “mixed together,” it becomes a car.

As much as a car won’t drive well if the wheel alignment is not proper, the same is true with music. You need to mix the sounds properly in order to get a good-sounding mix that plays well on the radio and other sound devices.

Audio mixing is not only used in music; it also applies in live stage performances, film, television, and video games.

The Most Important Tools You’ll Need

The proper tools you’ll need to get a good sounding mix are equalizers, dynamic processors (compressors, limiters, de-essers, etc.), audio effects, a mixer, and a good DAW.

You can choose to do all your mixing using software or outboard gear; it’s totally up to what you can or can’t access.

Technology these days is too advanced, so I believe it’s possible to get a proper-sounding mix using only software.

For me, it doesn’t matter what you’re using; just make sure that the music sounds good when compared with other songs.

If your audio recording is bad, then you won’t be able to fix it in the mixing stage.

Make sure all the sounds you have already sound good before getting into mixing.

Without a good mix, you also can't get a good master, no matter if you use the best mastering engineer in the world.

That should help you understand what mixing is, and in the next pages, I'll be showing you how to use all the tools to get a great-sounding mix.

BEGINNERS GUIDE TO MIXING

Mixing is definitely the most important part of the music production process. This is where you bring all your recorded sounds and samples together to mix them into a final song that a listener can enjoy.

This process involves shaping sounds by using tools such as EQ, compression, level balancing, stereo shaping (pan + mid side processing) and many other tools such as reverb, distortion etc.

At the beginning stages of the mixing process, you want to focus more on the overall level and tonal balance.

There are two tools you can use to achieve good balance in your music.

The first one is compression.

Use compression to control dynamics (ranges of volume from loudest to quietest parts of a sound or instrument), which will result in a well-balanced mix level-wise.

The second tool is EQ. An EQ is used to get a good tonal balance for the bass, midrange, and treble (high frequencies).

Both of these tools can be used creatively to achieve some great effects in your music arrangement as well. They're not just for overall balance.

Once you have a tonal and level balance that you're happy with, the next step is to add effects to create more space and movement in your mix.

This will create a three-dimensional space for your music. This involves using tools such as reverb, delay, chorus, phaser, panning techniques, and stereo enhancement tools.

After getting a good stereo image, your mix should be ready for mastering. Or you can take it to the next level by using analog summing.

Later in the guide, I'll show you how to do analog summing using Waves plugins.

You can also add some character, warmth, and punch to your mix by adding saturation and distortion during the beginning or final stages of the mix.

Don't worry too much if you don't understand some of the stuff I mentioned above. I'll talk more about how to use these tools later in the guide.

Right now, I'm basically giving you the bigger picture of what mixing is and why it's important.

Note that the tutorials in this guide are done using Cubase, but all the principles and guides mentioned are universal. That means they "EQUALLY" apply to any reputable music software (DAW).

GETTING STARTED WITH MIXING

This is where a lot of people fail.

Your first approach to a mix will determine if you're going to get great results or not.

Before you even open your DAW, you need to know what the final results should sound like. If you just dive right into mixing without picturing the final results, you're not going to get a good mix.

That's what I call "hope mixing." You just hope that adding a reverb will improve the sound, or that boosting 3 dB at 50 Hz on a kick will make it sound PHAT.

You need to avoid any guesswork as much as possible, right from the beginning.

But before I get into how you can avoid guesswork in your mixes, I need to share the biggest mistake that many mixing engineers make.

I didn't do mixing like this when I started, and that's why I was failing to get top-quality results.

I even taught people to make this mistake in the past (**not intentionally, of course**), which I strongly regret now.

Mix Preparation

A lot of people fail to get great mixes because they skip this part.

Preparing for mixing is really important; it makes your workflow a lot easier, and you'll be able to finish your tracks a lot faster. You need to be well organized, or else you run the risk of failing before you even start mixing.

The first thing you need to do is separate your production project from your mixing project.

You don't want to be working on your production while mixing.

Finish your production first so that when you go to mixing, you can only focus on engineering the song.

Bounce all your sounds into WAV or AIFF files, **not MP3**.

The next thing you want to do is color-code your sounds. You want to have all drums with one colour, guitars with a different colour, same applies for vocals, brass sounds, bass, strings etc.

This will help you navigate your mix 10 times easier.

You don't want to spend 2 minutes trying to find the snare drum while mixing; that will drop your energy and you'll end up frustrated.

You don't need that while mixing; you need to stay focused.

All DAWs make it easier to color code your tracks, so use that to your advantage.

The next thing you need to do is group all your sounds based on the instrument. Route/Send all your drums into 1 group, do the same thing for the vocals, guitars, bass, keyboards, synths etc.

You can also color-code your group channels to avoid any confusion.

Once you have all your group channels sorted, create a new bus or group channel and name it "mix bus" or whatever you prefer.

Route all your group channels to the mix bus channel and leave your master channel untouched. Do all the mix buss processing on this mix buss group channel; don't do it on the main output master.

This will make it easier to use reference tracks. Your reference tracks won't be affected by any processing you do on the mix bus channel.

You don't want to keep disabling the plugins in the master channel when comparing your mix to other reference tracks. This will save you a lot of time and frustration.

Sometimes you might forget to disable the processing on the mix bus, and your reference tracks will be affected, which will result in a poor comparison and lead you in the wrong direction.

So you need to avoid that by all means.

Also, make sure that the volume fader on your mix bus channel is always at zero; don't touch it. Treat it like you would treat the master channel.

After you've sorted your sounds, you need to add some essential effects channels.

A good starting point is to add three different reverbs (hall, plate, and room).

Next, you can add two different delays (ping pong and slap), then add one chorus effect and a distortion effect.

Simply add the ones you think you'll need in the mix.

Obviously, you're going to change the settings of the plugins later, but as long as they are there and ready for use, that makes your job easier and will make your workflow a lot faster.

Once you have all that sorted, you're ready to mix, and you can move on to the next module.

ONLY Use Buss/Group Processing

This is the top secret that pros keep from beginners. Top engineers never mix individual sounds; they always mix them in groups.

Of course, there are some cases where you have to work on an individual sound, but I encourage you to avoid it as much as possible.

The only two sounds you can mix individually on a song are the kick and the snare. That's it!

Please understand that I'm not talking about grouping different sounds or instruments.

If you have a stack of guitars, don't mix them individually; instead, group them into a bus channel and process them as one. The same applies if you have a stack of vocals.

Especially if the sounds were recorded with the same microphone, grouping them and processing them together will glue them together and make everything sound like one.

Even if the sounds were recorded with different microphones, processing will glue them together and make them sound like one.

You can separate the lead vocals from the backings. Same thing applies if you have lead guitars and a solo guitar; you can separate them and process them differently.

I know this can be hard at first, but trust me, it makes a huge difference and you start to look at mixing in a different way.

What I mean is that you start to focus on the entire context of the mix instead of individual sounds.

That is why this is so powerful, because you get to step into the shoes of the consumer.

Never Touch The Solo Button

Another thing that beginners fail to do is to avoid the solo button.

It's still tempting for me to solo something even today because I got used to using the solo button for a while when I was still learning how to mix.

So it's going to be hard for you as well at first, but make sure that you never solo any sound in your mix.

If you feel like you really need to solo something, then instead of soloing an individual sound or channel, solo the entire group or mute a few group channels that are distracting you from hearing the sound clearly.

Another thing you can do is increase the volume of the sound you want to solo. By increasing its volume, you'll be able to hear it more clearly.

Then, when you're done with the processing, you bring it back down again.

I encourage you to never use the solo button during mixing. Keep the spirit of being in the shoes of the end user.

Now we can move on to the next strategy about how to avoid guesswork or "hope mixing."

How To Avoid Guess Work During Mixing

Most people will never even realize that they rely on hope-mixing.

Before you get started with mixing, the first step is to find the correct key of the song.

If you're working on a client's mix, then ask them to give you the key of the song. If it's your song, you should be able to identify the key right away.

Once you know that your song is in the key of 'C' for instance, you tune all your drums and samples to the same key or at least in the scale of 'C'.

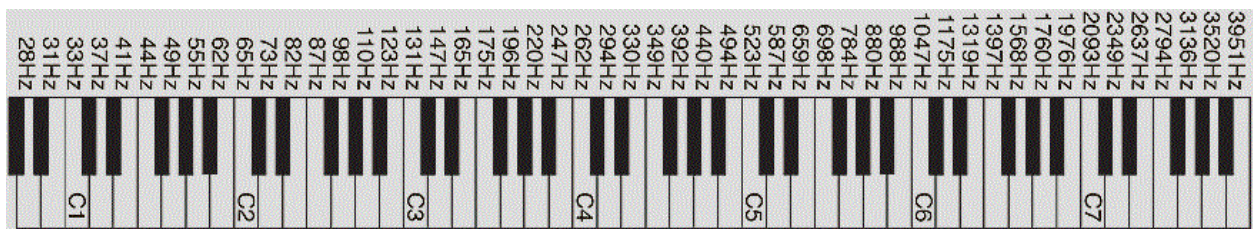
This is a music production and sound design technique, but most producers skip it. Basically, you need to change the pitch of the

sound (kick, snare, toms, etc.) to match it with the key of the song.

Use a frequency spectrum analyzer to see where the loudest peak of each sound is peaking at. For instance, if the kick drum's loudest peak is at 62Hz that means it's in the key of 'B'.

The 'B' key on a piano is 62Hz. So if your song is in the key of "A," then you have to pitch that kick drum sound down until the loudest peak hits at 55Hz.

The graph below will make it easier to know the frequency value of each key.



Most equalizers and frequency spectrum analyzers come built-in with this kind of graph to make it easier for you.

Let's say you want to have a kick that's in a different key. Play on the key of "C."

Another trick is to simply look at the graph above, then open a parametric EQ and create small "narrow" boosts at 33Hz, 65Hz, 131Hz, 262Hz, 523Hz, 1047Hz, and a final boost at 2093Hz.

Boosting all the frequencies I mentioned in one sound will force your sound to be in the key of 'C'. You can do that with any key; it doesn't have to be "C."

Another thing that I have to mention is that when tuning your drums or samples, you don't have to tune them to the root key of the song all the time.

In some cases, you have to use the scale of the root key.

So, for instance, if your kick's loudest peak is at 65Hz which is the "C" key on a piano, and your song is playing in the key of A major (the 'A' key on the piano is 55Hz).

Since 'C' is not in the scale of "A Major," you have to pitch down the kick from 65Hz to 62Hz which is 'B' on a piano, and 'B' is in the scale of "A Major."

You can also try to pitch it even further, to 55Hz ('A' key), which is the root key of the scale, and see how it sounds.

The kick might sound weird and change character if you pitch it too much. So your best option is to use the scale. That means you have to pitch your sound to any key that's in the scale of "A."

Or simply create a narrow boost for every frequency that is an "A" on the graph above to force the sound to be in the key of "A."

Once you have all the sounds in your mix playing in the same key or scale, then your sounds will automatically blend well together and there will be less clashing.

This makes your entire mixing job a lot easier.

This is why I don't teach music theory or production; this stuff is really hard to explain, but I hope you understand it.

Read it a couple of times to understand it if you're still confused.

For live drums, I recommend pitching them before hitting the record button; I wouldn't recommend tuning live drums in the mixing stage.

But if you feel like you have to, then go for it if it sounds like it doesn't mess up the timbre or character of the drums.

Now let's get back to the topic of avoiding guesswork in your mixes.

Getting your sounds in the right key is the first step, and don't take this lightly. It could be the only thing preventing you from getting pro-quality mixes.

Step 2 is getting eight to twelve reference tracks that are in the same key as your song. If they're not in the same key, you run the risk of boosting or cutting the wrong frequencies in your mix.

It will be hard for you to get a good tonal balance if you simply pick any random reference tracks that play in different keys.

It's OK if you can't get 12 tracks; just get as much as you can. I just find 8 to 12 tracks to be a sweet spot. That way, I'm able to find all the instruments that are in my mix in all these different songs.

That's how you avoid guesswork; you simply listen to how a sound was mixed in a particular song and replicate it. It's that simple; don't complicate it.

In most cases, I get to hear how different engineers mix the same sound and go with what will work best for my song.

Now before getting into how to properly use reference tracks, there's a sneaky little technique you need to use before you can reference your music.

Master Your Mix

Here's a sneaky technique that is used by professional mixing and mastering engineers.

I think this is a secret because I've never seen anyone teach it online. I always see this when watching the pros mix.

The trick here is to pre-master your song to get an idea of how the final results of the mix will sound.

In mastering, the engineer will add a limiter or maximizer first **(on the last insert slot of the chain)** to hear how the mix will sound at its final stage.

Then they'll start the mastering processing with the limiter still on. *Not all mastering engineers know this technique.*

The same technique is used in mixing by simply pre-mastering your song before you start mixing it. I say “pre” because this is before mastering, but you’re actually mastering the song.

Basically, you’ll process your mix bus channel. Here’s a simple chain you can use.

- ☐ Buss Compression
- ☐ EQ
- ☐ Distortion/Saturation
- ☐ Stereo Enhancer
- ☐ Limiter/Maximizer

The limiter always goes last in the chain; never put it in the first insert slot.

After using this strategy many times, I realized that in most cases, my songs didn’t even need mastering. But I sent them for mastering anyway, just to be sure.

The reason I don’t master my own music is because I’m not using a million-dollar studio, so I could miss a thing or two that the mastering engineer can fix easily.

Now, after doing the mastering, you can go on to the next step, and you’ll see how powerful this process is.

You’ll also notice that this was the missing piece in your mixes.

Make sure that you give yourself some headroom before mastering, and don't push the limiter too hard; give it some room to breathe.

You need to give it some room to breathe because you're still going to be adding processing tools to your mix, and those will increase the overall volume.

How To Use Reference Tracks Like A Pro

It's time to learn the power of using reference tracks.

Don't ever do this step without mastering your mix.

The reason we master the mix is to have a fair comparison of a master with a pre-master instead of comparing a mix with a mastered version of a song.

You can't compare apples with oranges, so you have to make your mix a master so that you can make a fair comparison.

WHY DO YOU NEED TO USE REFERENCE TRACKS?

For me, the answer is simple. I don't have a million dollars to build a recording studio that has perfect acoustics and fancy outboard equipment.

I have to spend money on the tools that I need the most, and a perfect acoustic treatment is not one of them.

I bet you can relate to that.

So using reference tracks will definitely help you get a professional sound with very minimal tools.

IMPORTANT TOOLS YOU NEED

Back then, you would load all your reference tracks into your project and keep switching from track to track, which was time-consuming.

These days, there are tools that make referencing easy.

The one I use and most recommend is a plugin called **REFERENCE**.

NOTE: Each time I use the word REFERENCE in capslock (capital letters), I know that I'm talking about the plugin, but if I use small letters, then I'm talking about reference material or songs.

You can download it here: <https://audiospectra.net/reference>

Other great reference plugins that I recommend are:

ADPTR AUDIO Metric AB: <https://audiospectra.net/adptr-metricab>

Tonal Balance Control:

<https://audiospectra.net/tonal-balance-control>

For me, REFERENCE is the best plugin for now. ADPTR AUDIO Metric AB is also great.

If you choose to use Izotope Tonal Balance Control, then make sure that you edit the song you want to reference. You'll need to cut the reference song and only use the loudest part for reference.

Or else Tonal Balance Control will compare the beginning of a song with your loudest part, and that won't be a good comparison.

If you're not using any plugins for reference, then edit the reference songs and only import the loudest part into your mixing project.

ALWAYS go to the loudest part of your song when mixing it, especially where all sounds are playing.

Loop that loud part and focus on it. Don't jump from one section of the song to another unless the sound you want to mix is not playing in the loudest part of the arrangement.

With REFERENCE and Metric AB you don't need to edit your reference tracks because you can navigate to the loudest parts of your reference track or loop a section within the plugin.

All three plugins will tell you if your song is tonally and level-wise balanced. Izotope Tonal Balance Control and REFERENCE will also

help you see if your song is over-compressed or still needs more compression.

Tonal Balance Control only shows you compression for the low end, but REFERENCE shows you compression for each frequency band (bass, mid-range, and treble). That's why I love it.

Metric AB and REFERENCE also show if your mix is too narrow or too wide, which really helps you get a great stereo image balance.

Like I said, no more guesswork 😊

When you use the REFERENCE plugin, you eliminate most (if not all) of the guesswork when mixing.

So no more "hope mixing."

How To GET THE MOST OUT OF THESE TOOLS

I hope by now you see how powerful reference tracks are for your mix.

Now, before I close this chapter, I just want to give you a few tips that have really helped me and made the whole process a lot easier.

You need to use one of these tools to make your mixing process 10 times easier. Remember that 80% of your mix is balance, and the other stuff is 20%.

That means when you have the right tonal balance (bass, midrange, and treble), the right level balance for your instruments, and the right compression, you have 80% of the mix done.

After you've grouped all your individual tracks on a bus channel and mastered your mix, you need to switch between all the reference tracks to see what needs to be fixed.

Make sure that you loop the loudest parts for both your mix and the reference tracks.

I encourage you to start with getting the right level balance before going into tonal balance and compression. It will be a lot easier if you approach it like this.

As you keep switching between tracks, you'll be able to hear what is loud or quiet in your mix.

You simply adjust accordingly. But before you increase volume, always see if there's something you can lower in volume to help the quiet sound be a bit louder.

What I mean is that since your faders will be at 0dB at this point, you want to focus more on decreasing volume than increasing volume because your mix will clip or distort.

A trick I use is to get all the faders at around -6dB and all the pre-faders at -6dB as well. Some DAWs don't have the pre-fader feature, so -6db on the individual channel faders should work.

That gives me an overall headroom of -12dB and some space on the mixer faders to increase volume if necessary while avoiding distortion or clipping the overall mix.

Don't go below -8db on your digital mixer fader for any individual channel; the fader will lose resolution.

Make sure that you do all that before you master your mix.

So as you keep switching between the tracks, you listen to what needs to be increased or decreased level-wise and adjust accordingly.

Once you're happy with the overall volume and how each sound is sitting in the mix, you can move on to tonal balance.

To get a good tonal balance, you need to be familiar with how an equalizer works. You have to understand the frequency spectrum and its parameters.

I'll go into detail about EQ later in the guide; for now, let's focus on how to use reference tracks.

As you keep switching between tracks, you should be able to hear which sounds in your mix sound harsh, muddy, too bright, etc., and then adjust accordingly till you get great results.

The REFERENCE plugin makes this ten times easier by visually indicating whether you have too much bass or too few high frequencies.

Use as many reference tracks as possible to compare how different engineers EQ the same sound and choose the one that will work best for your mix.

Once you're happy with your tonal balance, you do the same thing with compression. Again, the REFERENCE plugin makes it easy for you to see if you need more compression or less.

Izotope Tonal Balance Control shows you compression for the bass frequencies only.

For the other frequency bands, you're on your own.

If you have REFERENCE or Metric AB then the next step will be comparing the stereo image balance.

This is your left, right, and center channels. REFERENCE or Metric AB will compare your song with your reference tracks to see if your mix is too wide or too narrow.

Simply switch between tracks to see which song has the stereo image you like the most and adjust your settings accordingly using stereo shaping techniques.

If you don't have reference plugins, then you're going to rely on your ears (it's not hard). Just keep switching between the songs to determine if the stereo image of your song still needs fixing.

Once that is completed, you will have completed 80% of your mix.

For the other 20%, you're going to rely on your ears.

Keep switching between tracks to determine the amount of reverb, delay, saturation, chorus, distortion, etc.

This is much easier to hear. You don't need to be an expert to determine how much reverb was added to a vocal, guitar, or any other instrument. This applies to other effects as well.

After you're done with the 20%, you can remove the loop on your song and listen to it from start to finish, adjusting any obvious problems that you come across.

Remember that **your song doesn't have to sound exactly** like the reference.

It just needs to be as close as possible to the reference track, and the mastering engineer will take care of that 5% you might have missed.

Never Do Low Cut (High Pass Filter)

Both Grammy Award-winning mixing engineers I've watched mix music from beginning to end NEVER used a high pass filter **by default**.

They never low cut all sounds and instruments.

This is the number-one reason why your songs sound thin when compared to other songs.

I know there's an argument about this online, but I challenge you to do the comparison for yourself. I've tested both and found that a mix with fewer high-pass filters sounds more punchy and full.

On the other hand, the mix with high-pass filters sounded thin and weak. It also sounded harsh in the upper mid-range.

A low cut also changes the phase and correlation of a sound. This is why I advise you to only mix on group or bus channels.

Whenever you create a cut or boost on an EQ, it changes the phase a bit. So if you have individual drum sounds, it's better to EQ the drum group channel.

If you add an EQ to all of them individually and the drum bus, you'll have a lot of phase shift as compared to using just one EQ.

Every parametric EQ creates a small phase shift; the problem occurs when there are too many EQs in your mix.

So avoid low-frequency cuts; maybe you can use them on a hi-hat, room mics, overheads, and cymbals. But for the rest of the drums, avoid it.

It can also be used to eliminate "obvious" low-end rumble or noise floor from vocals, guitars, keyboards, etc.

Use it only on recorded material; if you rely on digital sounds, then avoid low cuts as much as you can.

The reason people recommend high-pass filtering is because they know most people use untreated rooms, so they won't be able to hear the low end properly.

But we're using reference tracks and a powerful plugin, so we can solve the untreated room problem easily.

Don't get it twisted; I'm not saying don't use high-pass filters at all. Just **don't use them by default**. Use your ears or a frequency analyzer.

The Digital Mixer

Now it's time to look into the most important tool for mixing, the mixer.

The main purpose of a mixer is to combine all the individual sounds that you have separated into different channel strips into one audio signal.

Each channel strip will have what is called a level fader, which controls the volume of the sound.

This might look like an easy thing to do, but getting your sounds balanced with the most important sounds upfront and the other sounds at the back of a mix can be somewhat challenging.

The key is to make the most important sounds (the main groove) sound louder. For instance, if you're mixing dance music, you know your main sounds are going to be the kick drum and the bass.

For pop music, the vocals should be upfront in the mix.

Simply learn your music and pay close attention to your favorite songs in the genre you make and replicate.

The best approach is to drag all the faders down to zero and bring them up one at a time, but bring them up in a particular order, mainly by the order of importance, so that you keep the main parts louder than the other sounds.

Make sure you loop the music; don't stop the music, just keep it rolling. Move around the room, keep opening and closing the door, and even listen to the mix from the next room if there's one.

Go to the kitchen and make some coffee while the music is playing in the other room.

Shut the door and listen from outside. Listen to the music from as many angles as possible. This will help ensure a well-balanced mix.

The next thing you need to look at is the headroom to make sure that your song is not clipping.

It is important to leave some headroom when you're mixing so that you don't exceed a certain amount of loudness.

Imagine if you were tall and you reached the ceiling; it would be really hard for you to move in the house, right? Similar concepts apply to your mixes.

A mix doesn't need to be loud; that's what mastering is for.

If you want to know how much headroom you need, just ask your mastering engineer, but most engineers will say -6dBFS. Make sure that you determine the headroom by listening to the loudest part of your song.

One other way to use faders is when you do gain riding, also known as volume automation.

You can use gain-riding to make some of your sounds become loud or soft in other parts of your song.

You can also achieve the sidechain effect by using gain riding, for those who don't know how to use a compressor. All this depends on the style of music you're making.

A mixer will also have what's called "inserts," which are used to add effects to individual sounds in your mix. But the inserts are not the only way to add effects; a mixer will also have what's called "auxiliary channels."

Aux channel strips are used for what is called "parallel processing," which is mixing a dry signal with a wet or affected signal.

Parallel processing saves a lot of CPU usage as compared to adding effects to each and every channel in the mixer.

For instance, you can have three different reverbs in different aux channels. A room reverb for your drum sounds, a plate reverb for the vocals, and a hall reverb for other instruments.

Mixers can have a lot of features, like an equalizer and even a pan parameter on each channel strip.

Some mixers even have some kind of advanced routing feature whereby you can send one channel's output signal to another.

Normally, a channel's output will go straight to the master channel, but other mixers will have bus channels, also known as group channels in other DAWs.

These help you group certain sounds in a mix into one audio output.

This is useful for grouping all your drum sounds to tighten them up and play them together as one, which can also make them punchier.

Finally, all mixers will have a master channel strip, and this is the final destination for all the audio signals in your mix.

There's no reason to move this fader; the best thing to do is to keep it at 0dB.

If your song is clipping, then fix that using individual audio channels or group channels.

There's also no need to add effects in this channel; rather, do that on other channels as recommended above. Leave this as is, and the mastering engineer will take care of the rest.

Don't be afraid to open the manual for your DAW; it will give you a much better understanding of all the features, which will make your workflow a lot faster.

The manual will help you eliminate a lot of guesswork.

MIXING TIPS & PITFALLS TO AVOID FOR BEGINNERS

Rest Your Ears

Using reference tracks will help you mix your songs really fast. But make sure that you take a couple of breaks so that you can rest your ears.

Once you spend too much time on a mix, you'll get ear fatigue, and that's how you end up making wrong decisions. Even if the client is in a hurry, take a few 5- to 15-minute breaks to rest your ears.

Each time you come back to the mix, you'll notice a few things that need to be fixed. You do that until you've got nothing else that needs to be fixed.

Avoid Raising the Levels

Every time you feel like something needs to be raised in volume, first see if there's anything else that you can bring down in volume to help that sound (or sounds) be more audible.

This also applies to EQ, rather than immediately boosting it. First, see if you can reduce the frequencies of other sounds to make the quiet sound stand out in the mix.

So you must first think about what you can eliminate before you can add.

Test Your Mix

Once you're happy with your mix, always turn it up and listen to the mix from a different room. You must also listen to the mix from outside the house, apartment, or studio.

This will give you a broader perspective on your mix.

Do the same thing when testing your mix on different sound systems. For instance, during the car test, listen to the mix while inside the car with the doors closed and test again with the doors open.

Next, test the mix while standing outside the car with the doors closed, and then again with the doors open.

Also find different sound systems that are not in the room you're familiar with, maybe at a friend's house or if you have a home theater that's in a different room.

Test, and just keep testing till you get the best results. It's also a good idea to have your reference tracks with you while doing the tests so you can keep switching between the reference and your mix.

Avoid Headphones

Even though it's better to use professional studio monitor headphones as compared to using ordinary desktop or home

speakers, I would avoid headphones because they create ear fatigue way too quickly.

So it's very easy to make mistakes.

The problem is that the way they produce sound is not a natural way of listening to music.

What I mean is that when you're listening to music on speakers, your right ear will pick up the sound that comes out of the right speaker first, and then there'll be a small delay for the left ear to pick up sounds that are coming from the right speaker.

But the left ear will get the sound that comes from the right speaker as well. The same applies vice versa; the right ear does pick up sounds coming out of the left speaker.

So on headphones, the left ear is only fed the left channel of the stereo field, and that's unnatural.

Headphones can be useful during your mix testing stage and just for reference. You do need to use them to check for things like too much reverb, clicks and pops, unwanted noises, and panning.

Headphones can damage your ears, which is not something that you want since you work with music.

You don't want to be a mixing engineer with ear problems. So avoid headphones and earbuds as much as possible.

Love your ears; they're your best asset.

Don't get it twisted; I'm not saying you can't get a Pro mix on headphones.

Mix Buss Limiter & Compression

Since I recommend that you add a limiter and compressor to your mix bus, I have to warn you not to overdo it. Leave some room and let your mix breathe.

The good thing about using plugins like REFERENCE is that the plugin will automatically bring down the volume of the other songs to match your mix.

So you don't really need to go drastic on the compressor and limiter.

Less is more.

That's why I didn't recommend that you add a multiband compressor to the mix bus (the master channel for your mix); it would be overkill.

The goal of adding a compressor and a limiter is to help you find sounds that are obviously too loud or too quiet in your mix very easily. They'll stand out, and you'll be able to pick them up and adjust accordingly.

Basically, this helps you get a really good overall balance.

Don't Touch The Master Fader

If you're a semi-pro or pro, you might be thinking that's obvious.

But beginners always make this mistake.

Never touch your master fader. Keep it at 0dB.

Use your speaker's volume or audio interface to add or decrease volume. Most software these days comes with a control room; use it to your advantage.

Use Fx Channels For Effects

Never use effects such as delay, reverb, chorus, etc. as an insert on an individual channel.

Use FX or Aux channels for your effects.

You want your drums to have one reverb, which will glue them together nicely as compared to inserting different reverbs on each drum sound.

Inserting different time-based effects on individual channels will make your drums sound as if they were recorded in different studios. They won't blend well together.

The same thing also applies for vocals, guitars, and other sounds.

You want to use one reverb for all your vocals and one reverb for your entire drum kit; the same goes for guitars and other sounds in your mix.

So using an FX channel will allow you to use one reverb and send it to your individual channels, which will also save tons of CPU usage.

This also allows you to blend the dry signal with the reverb instead of making your sounds swim inside a reverb or delay.

This gives you flexibility to personalize your effects.

Don't Over-Mix

Spending too much time on a mix is not always a good idea; you don't want to smooth off all the edges.

Once all the sounds are clean and working well together, you can be overly excited and mess up the overall balance.

The aim is not to make it super clean and perfect; just get it done and ready for mastering.

Or else you run the risk of destroying a good mix.

A/B Test All Processing

Each time you add a processing tool such as reverb, EQ, compressor, etc., make sure that you A/B test the final results. You must always check before and after.

Also, make sure that you don't use the solo button when you A/B test; always listen to the processing in context.

If you can't hear the difference properly, boost the volume of that sound a bit, and then do your A/B testing. After you're happy with the results, you can bring it back down again.

Beware Of Volume Changes When Processing

After processing a sound, always check that the volume didn't change.

Most of the time, beginners can confuse something that sounds good with something that just became loud.

This happens when you add any processing tools.

Some plugins will add a small amount of saturation even before you start tweaking them. As you keep tweaking the plugin, you'll add more volume.

So make sure that when you do your A/B test, you also check that there's no change in volume. In most cases, there'll be some change in volume, so adjust the gain and output of the plugin to match the level of the dry signal.

All plugins have an output gain, I think 😊

But the ones I use always have a gain and I use it all the time, with EQs, compressors, reverbs etc.

This way, you get a good comparison of the results before and after. Some plugins come with a "Gain Match" feature; that is even better. Use it all the time.

How To Use Effects & Processing Tools Like A Pro

In this module, we'll be looking at how to use effects and processing tools like Pro Engineer.

I'll be showing you how to use these tools the right way so that you can improve your sound.

Audio processing tools can make or break your song.

They can add elusive power, punch, depth, and detail to your music or make your song lack interest and fall flat as soon as it drops out of the speakers.

These tools are very important for any genre of music, even in a classical piece.

In this module, we'll be looking at how and when to use these processing tools to take your mix to the next level.

These will also help you make your music more engaging and sound professional.

Let's start with the most common effect, which is delay.

Creative Ways To Use Delay During Mixing

The delay effect is very simple in design and simple to understand. By just tweaking all the knobs, you're able to hear what it's actually doing.

It's not complicated at all.

A delay effect records the input signal into a buffer and then plays it back after a certain period of time, depending on your delay time settings.

This creates a repeating sound, or decaying echo.

The delay effect is a really versatile tool and one of the most important tools to have in your Fx chain.

Effects such as chorus, reverb, flanger, and many other plugins are all made from the delay effect, so there's a lot you can do with it.

Delay can make a simple recording or a programmed instrument sound larger. It can also be used to widen sounds in the stereo field.

The delay is often used to create an echo effect, but it can also be used to correct timing issues and improve performance.

The echo effect sounds great when it's matched with the tempo of the song.

However, too much of it will push a sound back into the mix instead of being in your face.

The parameters of a delay will differ from unit to unit or plugin to plugin, but here are the most common parameters.

DELAY EFFECT PARAMETERS:

Gain: This allows you to set the signal level going into the delay.

Mix Knob: This allows you to control the dry and wet signals of the effect, which allows you to control the amount of delay you want to add to a particular signal.

LPF: This is simply a low-pass filter that lets you filter out some of the high frequencies from the delay.

HPF: This is a high-pass filter that lets you filter out the low frequencies from the delay.

Delay: This parameter controls the amount of time between the original signal and the repeated sound.

Depth: This parameter lets you add modulation to the delay so that you can create a chorus effect. It will help you to keep a sound upfront if necessary instead of pushing it to the back of the mix.

Rate: This setting lets you adjust the amount of time that the modulation takes to go one time through its cycle.

Feedback: The Feedback parameter controls how many times the echo repeats. A low setting makes the echo happen just once, and higher settings produce more echoes.

These are the parameters you'll find in most delay effect units or plugins. Many units won't have all these parameters, and you don't need all those parameters all the time.

Now let's look at how you can use this tool to improve your mixes.

50 to 100 millisecond delays are well known for the "slap back" double-track effect that was used in the 1950s; this adds an artificial double. This is great for busy vocals and when soloing with an instrument.

A delay that is in sync with the tempo of the song can disappear early or sound short, but this adds glue to the mix. This is something that the reverb effect cannot achieve; this helps your sounds work well together in a mix.

When the delay is in sync with the tempo of a song, it will create depth even if it's not noticeable, and it won't push the sound back in a mix.

Many delay effects come with a sync button, but if yours doesn't, you can use a simple formula.

Simply take 60 000 and divide it by the tempo of the song, and that should give you the correct delay time that will sync well with your song.

Example: $60\,000 \div 90\text{BPM} = 666.7$ milliseconds (1/4 note delay)

If that's too long, you can get the 8th note by taking 666.7 milliseconds and dividing it by 2, which gives you 333.35 milliseconds (1/8 note delay).

To get 16th notes, you take 333.35 ms and divide by 2, which gives you 166.675 ms (1/16 note delay).

The feedback parameter is used to set the number of repeats.

This can sometimes be called "regeneration" or "repeats," depending on the hardware unit or plugin you're using.

Longer feedback times tend to muddy up the mix and ruin the clarity of a sound. Use a short feedback time to keep the sound punchy.

Longer feedback is used if the music is slow and has less instrumentation, or as a creative technique.

To avoid the delay from sticking out in the mix, filter out the high frequencies on the return channel (delay signal). Most delay effects come with a filter built-in.

If your delay effect doesn't have filter features, then simply add an EQ or filter plugin to your Fx channel.

Filtering out the high frequencies on the delay signal helps your sound blend well in the mix; if it sounds exactly like the original signal, it might not work well, so always use filters.

The filter will soften the transients and push the delayed signal to the back of the mix, leaving the original signal in front.

Just like reverb, delay is mostly used in a send or return channel (aka FX channel) not only to save CPU but to have full control and flexibility.

EQUALIZING THE DELAY

A great way to make the delay signal blend well in a mix is to use an equalizer to shape the sound.

If you're mixing a busy track with lots of sounds, then EQ will be your friend if you choose to apply a delay effect.

Bright delays work well if the music is sparse, but for mixes with lots of sounds, use dark delays with more body and midrange frequencies.

Use the equalizer to remove low frequencies to avoid clutter in the low end. Do not shelf the low and high frequencies; filter them out completely with low-pass and high-pass filters.

How you shape the EQ depends on taste, what you might want to achieve, or the material you're working on.

Be creative and make sure that what you do benefits the entire mix, not one sound.

How To Use Compression On A Delay Effect

Most of the time, you will find that the delay keeps making the loud peaks jump in certain parts of the mix. But when you try to lower the amount of delay, the soft parts feel like they need more delay.

In that case, you'll need to add compression to the delay signal to keep its volume constant throughout the entire mix.

If you want to achieve a pumping sound delay, then you can use sidechain compression, also known as ducking delay.

It's a really cool effect if you don't want the delay to mess up the transients of the original signal.

Multiband compression can be used if you just want to treat a certain frequency band.

If drastic compression is still not taming out the loud peaks, then go for a limiter.

Compression will help you control the dynamics of the delay signal to keep your sounds clean and transparent.

ADDING HARMONICS TO YOUR DELAY EFFECT

Adding saturation or distortion to a delay effect can give the sound warmth, more color, and more character.

Most stock plugins that come built-in with a DAW don't add color or character to a sound. That can make the delay dull and boring.

In that case, you will need to add some harmonic excitement to make the sound rich.

You can achieve this by adding tape saturation, overdrive, or distortion in the delay Fx channel.

A subtle amount of saturation will work; you don't need to make it obvious or change the tone unless it sounds good, and that's the sound you're going for.

ADDITIONAL TIPS

- Rock and Metal vocals are mostly processed with delay instead of reverb because reverb tends to muddy up the vocals and push them at the back of a mix.
- A delay works well by affecting the signal without pushing it to the back of the mix. If you have a mix with lots of instruments or if the music is loud, then use mono delays for the lead vocals, especially in the verse.

- For the chorus part, add a reverb to the mono delay (you can automate this). This will make the delay wide and add contrast to your mix.
- To emphasize the width, simply add an EQ with M/S processing to boost the sides in the high frequency range. Add this right after the reverb in the delay Fx channel.
- Use a stereo delay for your backing vocals to differentiate them from the lead and push them back in the mix to keep the lead vocals upfront.
- If the music you're working on doesn't have a lot of sounds then you can use stereo delays with more feedback and long delay time.
- If a reverb is making any of your instruments sound distant, muddy or washy then switching to a delay will add the correct amount of depth you need without ruining the sound.
- Many times, one delay in an Fx channel can work for all the sounds in your mix. Sometimes you'll have to create different delay Fx channels with different settings.

Just make sure that you use an equalizer and saturation to help the delay signal blend well in the mix.

Using the same delay for most (if not all) of your sounds will glue the sounds together, make them sound like one, and help them blend nicely in the mix.

Delay can be used in a lot of ways and for many sounds. The only time I don't use it is when I'm mixing drums.

I "*might*" use it only on hi-hat, shaker, tambourine, cymbal, and percussion instruments.

If you want something that sounds different or unusual, then you can try the reverse delay effect.

Basically, you take a sound and reverse it (every DAW has a reverse feature). Then you add a delay effect to the reversed sound.

Bounce that reversed sound to a new audio channel. Then reverse the sound again, and when done correctly, the tail of the delay should be the attack.

That means you'll hear the delay signal before you hear the original signal.

This is a really cool effect that you can use in your arrangement to introduce different parts of the song, just like you would use a reverse crash.

Finally, be creative. Take these ideas and come up with something dope.

Creative Techniques For Using The Reverb Effect

The reverb effect is one of the most important tools in music production as well as audio engineering.

But the use of this effect can easily be overlooked or misunderstood by many, so for this tutorial, I'll share with you tips on how to use it effectively in your mixes.

A well-balanced mix with only EQ, panning, and dynamic processing can sound really good. But adding reverb just adds some flair and turns that good mix into a great mix.

In audio terminology, the "flair" is called a "dimension," which is the width and depth.

In this module, I want to show you how to use reverb in a different way instead of choosing a preset.

Great techniques that will help you design your own reverb, add some flair, and create your own reverb signature

But first, let's look at some common reverb parameters.

REVERB PARAMETERS

Room size/type: Whether you use a reverb patch within your DAW or a separate outboard reverb unit, you can choose the type of reverb that you want to use.

You have the option of a room, hall, or plate (a type of reverb that uses a metal plate to create the sound). As well, you can choose the size of the room in either meters or feet.

Decay: The decay is the length of time that the reverb lasts. Longer decay occurs in larger or more reflective rooms.

Predelay: The predelay is the amount of time from the sound's beginning to the start of the reverb (described in milliseconds). Predelay helps to define the initial sound signal by separating it from the reverb. This parameter is essential to making your reverb sound natural.

Density: The density parameter controls the level of the early reflections (the first few milliseconds of the reverb sound). This parameter enables you to simulate different sizes of rooms because, in a larger room, the main section of a reverb takes longer to reach you.

Diffusion: Diffusion affects the density of the reflections in the main section of the reverb sound. A higher diffusion setting results in a thicker sound.

With that covered, let's get reverberated.

Reverb is added to a sound to add width and depth, even if you have a really dry or dull sound.

It can also add excitement and make a sound bigger or wider.

If you want a bigger sound, then use a short reverb time or a long reverb to get the lush sound that you hear in soul music keys and deep house songs.

Reverbs are mainly used in an FX channel and processed in parallel with the dry signal for full control and flexibility.

When used in the signal's insert, you won't have as much control as when used in an aux or fx channel.

I'll show you how below.

Use reverb to push a sound further back in a mix.

For instance, if the drum kit is overpowering the vocals, you can use reverb to push the drums back in the stereo field so the vocals will be upfront.

So whenever you add reverb, remember that you're pushing the sound toward the back of the mix.

There are a lot of genres today, so listen to some of your favorite songs to figure out which sounds should be at the front and back of a mix and how much reverb is added.

You can use two simple formulas to get the correct reverb time for both the tail and the pre-delay.

To find a good reverb time that will be in sync with the song, simply take 60 000 and divide it by the tempo of the song.

Example: $60\,000 \div 120\text{BPM} = 500$ milliseconds (0.5 seconds)

The pre-delay formula is a bit different, but it's almost the same thing.

Example: $7500 \div 120\text{BPM} \div 2 = 31.25$ milliseconds (0.03125 seconds)

To make things easier, you can use these formulas to calculate your reverb time, or you can simply use your ears.

But using the formula will guarantee that the reverb is in sync with the tempo of the song.

Now, let's look at some techniques you can use to design a reverb that will fit well in a mix.

EQUALIZING THE REVERB

It's time to add that flair I talked about earlier. The first thing we're going to add is the equalizer.

The secret is to add the EQ before the reverb in the FX insert chain. That way, you'll be equalizing what's going into the reverb rather than what's coming out.

But test both ways and choose what you like; remember, you're designing your own signature sound and doing what's best for the mix.

What we'll be doing here is simply removing frequencies that might get in the way of other sounds to avoid masking.

The low-end frequencies should be removed as a starting point.

Also make a cut with a wider Q in the low-mids to avoid the reverb adding any mud and a wide Q boost from 10kHz to 12kHz to make the reverb bright and add some air.

A reverb always sits well in a mix when it's equalized. Play around with the EQ till you find settings that work well with your song.

DISTORTED REVERB

Adding distortion, saturation, or overdrive to a reverb can add some crunch, warmth, and personality.

This will also thicken up the reverb and give it that old tape or analog sound.

Make sure the distortion is really subtle; you just want enough to make the reverb warmer. Normally, anything between 18% and less will be enough.

A bit-crusher can also work well for this type of effect.

Just keep it as minimal as possible because you don't want to add too many harmonics, which will add up in the mix and create mud in the midrange or add harshness in the high frequencies.

THE THREE REVERB SETUP

This setup allows you to place all your sounds in different environments to give your mix more depth and excitement. You can achieve this by using a simple 3-reverb setup.

It's easy to use and a great technique for those who don't really understand reverbs.

Simply set up three FX channels: one with a short room reverb, one with a medium plate or ambient reverb, and one with a long hall reverb.

Each time you want to add reverb to a sound, you simply audition all three and choose the one that works best for the sound without messing up the entire mix.

Really simple and very effective.

WIDENING THE VERB

You can also add stereo image effects to make the reverb wider or make it play in mono, which will make it tighter.

You can use a stereo image processing tool to widen the entire reverb, or you can use mid-side processing to affect only the sides or mid channel.

For instance, if you just want to make the high frequencies of the reverb wide, you can simply boost the sides in the high frequency range using an EQ with M/S (mid/side) processing.

This will boost the clarity of the reverb effect. Or maybe you just want to remove the low frequencies of the reverb in the mid (center) channel, then use M/S processing.

Even stereo image tools can make a reverb fit well in the mix a lot better than choosing a preset.

Play around with it till you can make it sound good for your song.

REVERB AUTOMATION TRICKS

Sometimes you might find that the reverb is working well in the soft parts of the song or that it's just too much in the louder parts of the mix.

That's when automation comes in handy; simply automate the FX channel until it works well on the whole song.

Even if there are certain notes or loud peaks that keep jumping up in the mix, then use automation.

You can also use a compressor instead of doing manual gain riding to smooth out the reverb signal in louder parts.

One other trick that I don't use is the reverse reverb technique, which is mainly used on vocals. But it can also sound good on guitar solos and percussion sounds.

I've already explained how to reverse the delay effect above; it's a similar process. The only difference is that you'll be using a reverb instead of a delay effect.

Another neat trick that I don't see a lot of people talk about is the sidechain reverb technique.

Basically, what happens here is that when the original signal is playing, there will be no reverb, and the tail of the reverb only kicks in when the signal fades out (this depends on your compressor settings).

However, it produces a really cool pumping sound effect. Just keep it subtle and don't make it too obvious.

I just hope this tutorial has given you some ideas about using reverbs creatively.

Take these and add your own imagination to them so that you can create your own reverb signature.

Keep other engineers wondering: how did he do that?

Happy reverberating 😊

Using Distortion Like A Pro

Most people simply add a distortion and hope that it sounds right, so in this module I'll help you avoid the guesswork and show you pro secrets that will add more depth and warmth to your mixes.

In simple terms, distortion is clipping a signal to generate new harmonics for the sound that you're distorting.

I'm talking about good distortion here, not clipping or overloaded sound signals.

Just like running a clean bass tone through an amplifier or distortion pedal to create distorted tones that add warmth and harmonics to the bass.

This is a great tool for modern recording because digital tools tend to produce a clean sound that sounds "cheap" and lacks weight.

Modern plugins and technology make it possible to emulate all kinds of saturation, overdrive, preamps, amplifiers, and distortion "circuitry" within a DAW.

THE COMMON DISTORTION TYPES & HOW THEY'RE USED

To be honest with you, it is always better to use a hardware unit when it comes to distortion, overdrive, and saturation.

Those old units add a good texture to a sound that even digital emulators can't reproduce. So if you can get yourself a couple of those, then use them by all means.

They tend to be expensive, but they work better than digital emulators. Even if you get it secondhand, you'll thank me later.

I'm not talking about cheap guitar pedals here; I'm talking about units such as the **Culture Vulture**, **Sherman Filterbank 2**, or **Presonus ADL 600**.

Those are the ones you'll usually find in most top-end recording studios, and boy, they sound awesome.

Now let's get started and check out different types of distortion.

Tape Saturation

That "warmth" everyone is hunting for entirely comes from tape saturation.

Unlike other types of saturation, this one can be added to almost every sound and still sound good.

It combines transient smoothing, compression, and soft signal distortion to glue sounds together and add some Phatness.

Emulations often apply EQ, simulated tape hiss, and flutter.

Excellent for adding warmth and "glue" to a drum bus, vocal group, and synthesizers.

Valve Saturation

If you're looking for a deeper and richer sound, the valve is a great choice.

The valve adds odd harmonics in a musically pleasing way, with distinct coloration that makes the sound more aggressive.

Triode valves add even harmonics to a sound, while pentode circuits give odd harmonics and tend not to sound musical.

Awesome for adding color or utterly destroying existing tracks, buses, or a full mix.

Tube Distortion

Tube distortion sounds great because of the euphonic distortions it adds to the music.

Not only is the tube distortion harmonious, but it increases as things get louder, which helps a lot when you want to take a good musical performance and make it sound great.

This is mostly used in rock, metal, and other loud music.

Fuzz Distortion

Fuzz was the first type of distortion to appear in pedal form and was originally designed to sound like your amp was faulty or your speaker was damaged.

It is a very strong clipping that is usually used as a guitar effect. It adds harmonic distortion and massive sustain.

Those are the four most commonly used types of distortion, but there are more out there, and people are always experimenting with new stuff.

Parallel Distortion

Many distortion units come with a dry/wet parameter, which is really handy for creating parallel distortion.

This is due to the fact that distortion will add a lot of volume to a signal's input as well as the output.

The waveform of the original signal will also be clipped, shaped, or otherwise altered. So the distorted signal will sound louder than the original signal.

Mixing the two signals with parallel processing always yields excellent results. This makes the sound more audible on headphones and consumer speakers without destroying the signal.

But parallel distortion can result in what's known as the "comb-filtering effect." This is when there's a slight time/phase offset between the dry and wet signals.

It can be a pleasant sound, but to fix it, you can use all-pass filters, which will reduce the problem.

Another good trick to fix comb-filtering is to add a phaser effect before the distortion effect in your Fx channel. It will produce a less overtly phased sound.

Multi-band Distortion

Using a multi-band distortion will help you beef up certain frequencies of a sound. Like adding warmth to the midrange of a bass or vocal group.

Even if you have percussion or drum loops but only want to enhance the upper mids, you can use a multi-band distortion.

Instead of just one sound, multi-band processing always works better with a group of sounds, an entire mix, or for mastering.

Since you won't be using compression on a sine wave sound, a multi-band distortion is the best choice because it also adds a bit of compression to the sound.

Vocal Distortion

Vocals can also benefit a lot from distortion. But make sure that you use parallel distortion with an all-pass filter (low-pass and high-pass) so that you only add grit to the midrange without affecting the top and low-end.

Using all-pass filters will help you avoid making your vocals lack body and sound harsh and unlistenable.

Choose an overdrive distortion effect and use it as your first insert in your chain so that you can control the distortion with EQ and compression later on.

That will also help you achieve a fuller, more authentic sound.

Additional Tips

- Feel free to automate your distortion in certain parts of the arrangement if necessary.
- Distortion will help your sounds cut well through the mix and sound good on small speakers and ear buds. But too much will make your song harsh and unprofessional so take it easy.
- That's why it's always a good idea to use it as the first insert before using other processing tools so that you can control it with EQ.
- To avoid messing up things don't use it on individual sounds and end up with multiple distortion effects in your mix. Use it in parallel or on your group channels.
- Also avoid using it on the mix bus, let the mastering engineer deal with that. Distortion can easily make your mixes muddy.
- Check that you're not killing the dynamics of your sound, especially the transients or attack. It's easy to get carried away and mess up the transient and wonder why your song is not punchy.
- Use it if a sound sounds too thin, to add more depth for a sound that is at the back of a mix but want it to be upfront.
- Finally, avoid the clipping or crackling sound; your distortion doesn't have to be obvious.

How The Pros Use Dynamic Processing

Music would be boring if there were no dynamics (variations of loud and soft parts). Dynamics are what make the music interesting and keep the listener interested.

Dynamics are basically the loud and soft parts of a sound, mix, or arrangement.

For instance, when a vocalist sings certain words that are loud, it may be to emphasize what he or she is saying or to introduce the next part of the song.

Or a drummer playing softly in the verse and then starting to play hard to introduce the chorus.

Those are good dynamics and are really necessary to keep a song very interesting.

Then there are dynamic problems that you don't need. A good example is when a bass player or drummer can't play or hit all notes at a constant volume where required.

Those changes in volume will definitely need to be fixed so that no sounds end up clashing in the mix.

If you just leave unintentional dynamics in a mix, they'll pile up and make it hard for you to get a good balance for your song.

This also applies to problems such as background noise.

You don't want to end up with an unclear and cluttered mix.

That's when dynamic tools such as compressors, expanders, noise gates, transient shapers, de-essers, and limiters come in handy.

Unlike effects, these processing tools can be hard to understand for beginners.

Take me, for example. I graduated from music production school (2009) without fully understanding how a compressor works. I had an idea and knew what it did to sounds, but it took me two years to fully grasp it.

Basically, while I was studying, I was relying on guesswork. So don't feel bad if you don't get it right the first time; it will make sense as you keep practicing and using it.

The purpose of this guide is also to help you get a grip on it as fast as possible. I don't want you to wait 2 years to know if you're compressing correctly or not.

The Basics

Since there are differences in level between a sound's loudest and quietest parts, our goal is to change those differences by enhancing the quiet parts, lowering the louder parts, or removing the quietest parts altogether.

Basically, when doing dynamic processing, you're manipulating volume levels in real time.

COMPRESS TO IMPRESS

Let's start with the compressor, and we'll get to the other dynamic processing tools in the upcoming pages.

Compression is the process of shaping the dynamics of a sound.

It is essentially automated gain riding.

In contrast to reverb or delay, which are obvious to hear what they do to sounds, a compressor can be difficult to learn, and the effect it has on sounds is not easily audible to an untrained ear.

Just like EQ, you need to know why you are adding compression to a specific sound.

If you insert a compressor into a sound, you should have already pictured the outcome in your mind; if not, I would advise you not to use it.

Avoid compressors as much as you can; that's my tip for you.

If you program a kick drum using your mouse with the volume of the whole kick drum arrangement being the same throughout the entire arrangement, then that kick drum might not need to be compressed.

A compressor is an automated volume controller. It works well on sounds that have dynamics, meaning they change in volume over time.

A good example would be someone playing live drums. You know that person won't play the entire song at the same volume, so you use a compressor to keep their performance constant without the volume going up and down.

This helps the sound that is being compressed sit well in a mix.

But be careful, because too much compression will make your music sound unnatural, so don't compress everything and always think about the end results.

When compressing, you also need to think about the attack and body of the sound, as well as presence and punch. It is up to you if you want more attack for your sound or more body.

Giving your sound more attack will make it punchy, but that will reduce the presence and loudness of the sound.

On the other hand, giving your sound more body will increase its loudness and presence without allowing it to be punchy.

That's why it's important to picture the end results before compressing.

Don't always rely on a compressor to deal with dynamics, because sometimes the most effective thing you can do to achieve the same results or better is to use gain riding (volume automation).

DAWs these days make it easier to use manual volume automation, and you can edit a waveform visually.

Just find the easiest way to get the job done.

A compressor can ruin or distort sounds if not used properly; always keep that in mind.

UNDERSTANDING COMPRESSOR PARAMETERS

In this section, I'll be covering what role each parameter plays. It is crucial to know what each knob does before tweaking it.

Threshold and Ratio

A threshold determines at what level the compressor will start compressing the audio.

Any audio signal below the threshold level will not be affected; it will pass through uncompressed, and anything above the threshold will get compressed.

How much the audio needs to be compressed will be determined by your ratio settings.

A sound that is 6dB above the threshold with a ratio of 2:1 (which is half) will have its volume reduced by 3dB.

Ratio simply means how much volume will be reduced once the audio goes above the threshold.

The higher the ratio setting, the more compression will happen.

Attack and Release Parameters

A compressor also has envelope settings, which determine the speed it takes for the compressor to kick in and when it goes to rest. These are your attack and release parameters.

If there was no attack parameter, then the compression would kick in very fast and cause the sound to distort.

The attack is the speed at which the compressor needs to react when the threshold has been exceeded. A lot of people make the mistake of saying the attack is the time it takes for the compressor to kick in.

That's misinformation.

It's actually the speed that determines how fast or slowly the gain reduction must take place.

A fast attack will make the gain reduction happen a lot quicker, while a slow attack will make the gain reduction happen slowly and gradually.

The release is pretty much the opposite of the attack; it controls how quickly or slowly the level must return to being uncompressed once the signal falls back below the threshold.

To avoid the sound going back up to its original volume rapidly, you use the release parameter to smooth out the resting part of the compression and make the sound gradually go back to its original volume before the next hit (or note).

Here's a guide that will help you understand what is meant by "slow attack and release" or "fast attack and release."

Attack:

Super-fast: below 1 ms

Fast: 0.1 ms to 10 ms

Medium: 10 - 30 ms

Slow : 30 - 100 ms+

Release:

Fast: 50 - 100 ms

Medium: 100 - 200 ms

Slow: 200 - 500 ms+

Makeup Gain

Another parameter is the Makeup Gain.

Because the compressor reduces all the loud parts and leaves out the soft parts, you use the makeup gain to bring up the soft parts so that they can be equal with the loud parts and keep the volume of the sound constant and equal throughout the whole song.

Understanding Knee

The knee parameter determines the curve of the compression and how smooth or hard the compression must be when the threshold starts kicking in.

A hard knee means that all the audio signals that go above the threshold will be compressed equally with the same ratio setting. A hard knee will give you surgical control over the transients.

A soft knee allows the ratio settings to increase gradually, which will result in a more musical feel. So if your compressor is giving you an undesirable pumping effect, switch the knee to soft.

The knee parameter will need you to play around with it so that you can understand it, but it basically shapes the transition between the compressed and uncompressed signal.

There are no certain rules when it comes to choosing a hard or soft knee; this is more of an artistic choice.

Peak and RMS

There are 2 ways to measure the input level, which are peak and RMS.

Peak measures the level at a specific moment in time. Then RMS is the average energy level; it doesn't rely on transient levels like peak.

The compressor will react differently depending on what input level settings you're using.

In peak mode, every loud peak that goes above the threshold will be compressed. This is useful for precise dynamic control and for taming out loud peaks in an audio signal.

In RMS mode, the compressor will react less aggressively to the peaks, resulting in more natural and musical compression.

But all the loud peaks will be left unaffected.

To determine which one you must use, it will depend on your end goal and what you want to achieve.

The question you should ask yourself is whether you want to control the dynamics or have a more natural sound.

COMPRESSOR OR EQ FIRST

Another reason for you to picture the end results is because of the famous question of whether the EQ or compressor comes first in your insert chain.

You have to remember that an EQ can change the dynamic structure of a sound, and a compressor colors the EQ when used before equalization.

But this doesn't matter much if you're dealing with a subtle amount of EQ. This depends on the situation at hand.

Sometimes compressing a sound will not help but will make it worse; in this case, an EQ should be applied first, followed by compression.

Let's say you have a vocalist who has a lot of energy and is always kicking the microphone stand.

That will create some unwanted low-end frequencies, and if you compress the vocal, you just bring up low-end rumble (the mic stand noise).

So instead, you first cut out those low frequencies with an EQ, and that will help the compressor do its job easily.

Sometimes you might want to control the dynamics first before you equalize the sound.

Two Simple Tips to Guide you Through

1. EQ before compression if you need to compress drastically. If you don't need to EQ that much but you need to compress a lot, then insert your EQ before the compressor.

2. Compress before EQ if you need to EQ heavily; this is the opposite. If you really need to EQ your instrument to make it sound better, then insert the compressor before the EQ.

Why?

If you insert the EQ before the compressor, the compressor will compress your EQ fixes more than the rest of the signal.

Think about it: you're boosting certain frequencies to make them louder, and then you go ahead and compress them down again.

Seems redundant doesn't it?

THE RIGHT COMPRESSOR FOR THE JOB

A big problem beginners face is deciding which compressor to use for each instrument or sound.

Here are a few suggestions based on what the pros use. Even though the pros use real hardware compressors, Waves Audio has created some excellent plugin emulators.

You are free to use other company emulators, but I prefer Waves Audio because it is what I can afford.

If you can get UAD plugins, then go for it; they really sound good.

Kick Drum: dbx® 160, SSL Channel

Snare: dbx® 160, CLA 76

Toms: SSL channel

Overhead Mics: CLA 76, SSL channel

Room Mics: SSL channel, CLA 76

Bass: CLA 2a, dbx® 160, PuigChild

Acoustic Guitar: CLA 76, CLA 2a

Electric Guitar: CLA 76, CLA 2a, dbx® 160

Piano: CLA 76, PuigChild, API 2500

Vocals: CLA 76, dbx® 160, CLA 2a, PuigChild

Drum Buss: FabFilter Pro-C, SSL channel

These are the go-to compressors the pros use all the time, and I've never seen any of them break these rules. For each instrument, they always know which compressor to choose.

This doesn't apply to any other processing tools, but for compressors, it does apply because each compressor has a certain character and feel.

COMPRESSOR SETTINGS FOR DIFFERENT INSTRUMENT OR SOUND

Vocals: Fast Attack, Medium Release, Ratio 4:1, Auto Makeup Gain, Gain Reduction -4dB to -6dB, (VCA, FET, OPTO, VARI-MU)

Snare Drum: Fast Attack, Super-Fast Release, Ratio 4:1, Auto Makeup Gain, Gain Reduction -4dB to -10dB, (VCA, FET)

Drum Overheads: Medium/Slow Attack, Fast Release, Ratio 8:1, Manual Makeup Gain, Gain Reduction -6dB to -8dB, (VCA, FET, OPTO)

Kick Drum: Medium Attack, Medium/Fast Release, Ratio 4:1, Manual Makeup Gain, Gain Reduction -4dB to -6dB, (VCA, FET)

Bass Guitar: Medium Fast Attack, Medium Fast Release, Ratio 8:1, Manual Makeup Gain, Gain Reduction -3dB to -6dB, (VCA, FET)

Acoustic Guitar: Fast Attack, Medium Release, Ratio 4:1, Manual Makeup Gain, Gain Reduction -2dB to -6dB, (VCA, OPTO)

Picked Electric Guitar: Fast/Super-Fast Attack, Fast Release, Ratio 8:1, Auto Makeup Gain, Gain Reduction -6dB to -8dB, (VCA, OPTO)

Lead Synth: Medium/Fast Attack, Fast Release, Ratio 4:1, Auto Makeup Gain, Gain Reduction -4dB, (VCA, OPTO)

Smooth Mix Buss: Medium Attack, Fast Release, Ratio 4:1, Manual Makeup Gain, Gain Reduction -4dB to -6dB, (VCA)

Edgy Mix Buss: Medium Attack, Auto Release, Ratio 2:1, Manual Makeup Gain, Gain Reduction -2dB to -4dB, (VCA, OPTO VARI-MU)

These settings are just guidelines, so make sure that you experiment to find what will work best for your project.

The truth is that each type of compression will have a distinctive sound, and one of the secrets to achieving texture is having the right combination of compressors doing the right things.

That's why it's important to know when to use VCA, FET, OPTO, or VARI-MU.

VCA stands for "Voltage Controlled Amplifier," and its compression behavior is based on Peak, with fast attack and release.

Opto uses photocells as detectors and a light bulb to determine the gain reduction. Depending on the intensity of the signal as it passes through the light bulb, it will glow more or less.

Variable Mu aka **Tube** compressors produce incredibly smooth compression. Its transfer curve is far from being linear. The actual ratio rises as the gain is reduced. That means that the louder a transient is, the harder it is going to be compressed.

FET is usually faster than the fastest attack time on a Variable MU. If you are looking for transient control, this is the compressor type to go for.

USING PARALLEL COMPRESSION TO ENHANCE SOUNDS

This is a topic that I don't see a lot of people talk about, and honestly, this technique doesn't come to mind when I'm mixing a song, and it's been a while since I used it.

Don't get me wrong, I do use parallel processing all the time for effects, not for dynamic processors.

One thing I learned from Chris Lord-Alge is that digital parallel compression doesn't sound good. He doesn't use it with digital plugins; he only uses it with outboard gear.

So that's my reason for not using this technique. But I'll share it with you so that you know and understand it.

Parallel compression **sounds great when applied correctly.**

For instance, Grammy Award winner, recording engineer Michael Brauer uses parallel compression on vocals.

He sends vocal parts to different compressors using FX channels and blends them together.

In a 2013 Q&A session, Michael Brauer said he's not sure if this technique will sound as good when doing it in-the-box (using a DAW), but **it works fine in the hybrid.**

But in 2022, he released a series of videos where he taught the technique. You need to check it out if you're into parallel compression.

What is Parallel Compression

To put it in its simplest form, this is mixing a dry signal with a processed signal.

Sometimes the original signal is not 100% dry; it might have some subtle compression applied to it. But the processed signal needs to be heavily processed.

This will bring up the softest parts of the sound, adding audible detail while leaving the loud transients intact. *Hmmmm, that's rocket science.*

Put it this way: a normal compressor does what is called "*downward compression*," which is bringing down the loud peaks, while parallel compression is the opposite.

The quieter parts are brought up in level while keeping the other parts the same, so this processing technique is called "*upward compression*" and does the opposite of a normal compressor.

This technique is also known as New York compression.

You can do parallel compression by using send or FX channels.

In the digital domain, I would duplicate the sound to a new channel; this way, you'll have full control over both dry and compressed signals.

Some compressors come with a dry/wet parameter, so you can use that as well.

Parallel Compression Settings

Remember that **this is a concise guide** so the compression settings can be applied to any reputable DAW, Cubase, Ableton, Logic, FL Studio, Reaper etc.

When it comes to the envelope, you want to use a fast attack but don't kill the transients; a medium release will work best.

The ratio settings need to be high, which is why other people choose to use a limiter instead because it has an unlimited ratio.

Be careful; sometimes a high ratio can cause the signal to distort or create a pumping effect. While a small ratio of 2:1 may work in some cases, it doesn't work for this purpose.

You'll also need to use a high gain reduction, but this will depend on the material, so play around with a gain reduction of around **-10dB** to **-20dB** and let your ears be the judge.

Choose very high settings so that you can have full control of the two signals. But just make sure the two signals blend well when mixed together and don't make it sound obvious.

Keep everything as organic as possible.

Applying The Technique

Parallel compression can be applied to anything that you think needs upward compression; it could be vocals, drums, or even an entire mix during mastering.

As mentioned above, people like Michael Brauer use parallel compression on vocals.

If you're using this method on drums, copy the original signal to a new channel. If the drums are on separate channels, then export all the drums and import the drum track to the project.

Then add an EQ to the duplicate channel. You'll use a low-shelf filter to boost the low-end and a high-shelf filter to boost the high frequencies.

The EQ settings will be a **smile curve**.

After adding that EQ, add the compressor and use the settings I mentioned above. Once you're happy with that, drag the volume

of the processed channel back down and bring it up slowly to blend it with the original signal.

You can use this production technique on anything; just make sure that you **switch on the delay compensation** on your DAW to avoid phase.

Use the FabFilter Pro Q for these kinds of situations because it has both zero-latency and different phase processing modes.

You can add whatever you want to the processed signal chain; it doesn't need to be the EQ and compressor only.

Feel free to add your own creativity.

You can also make your mix punchy using parallel compression.

The question I see popping up a lot is, *"How do I know if everything is sounding right after doing New York compression...?"*

The simplest answer is: What were you trying to achieve?

If you just add something and hope for the best, then you're going to have a big problem. You should be able to picture the end results before doing any processing.

If you know why you added the compressor, then *you'll know if it sounds right or not.*

OTHER USES OF COMPRESSION

Another use for compressors is sidechain compression (we'll get into detail about this later on).

Sidechain compression causes the input signal to get in the way of the sidechain signal.

Typically used to make room in a mix for kick and bass. It is mostly used in modern dance music to create a pumping effect.

I've also met some engineers who use it on vocals to remove sibilance.

The input signal is sent through an EQ, and the sidechain compressor will trigger the equalizer gain whenever there's sibilance on the vocal performance.

However, this is something you can do easily with a De-Esser dynamic processor 😊

Then, lastly, we have multiband compressors, which are used to shape dynamics using a frequency spectrum.

These are typically used to compress a group of sounds rather than a single sound.

However, if you recorded a guitar with a sharply plucked note, you could use a multiband processor to compress only the high-frequency signal while leaving the rest of the frequencies untouched.

Multiband compressors can change the character of a sound because they work like an EQ as well, but when used correctly, they will change the sound in a good way.

If you know how to use a compressor, don't stress about using a multiband; they work the same way. The only difference is the UI (user interface).

Multiband compression is also used to fix inconsistencies in frequencies.

A good example of this is when a vocalist keeps turning their head left and right while recording. This will slightly change the frequency response of the vocal.

You can use a multiband compressor to fix those inconsistencies.

THE LIMITER

A limiter is a special type of compressor that will never allow the input signal to exceed a set threshold.

It is the same with a compressor, but with a fast attack, usually a hard knee, and an unlimited ratio.

Most compressors can only react to audio at the time it arrives, and limiters have a look-ahead feature that allows the limiter to see a few milliseconds before the compression takes place.

A normal compressor can squash a sound too quickly, which may cause it to distort, and a limiter might come in handy in that situation.

You can use limiters for a lot of different things in a mix. Even if you want extreme compression, you can use a limiter instead of a normal compressor, and it will sound smoother.

Or maybe you just want to remove some loud peaks, in which case a limiter can be a good go-to processor; it's even easier to use and understand as compared to a normal compressor.

There are also multiband limiters, which are mostly used in mastering to make a song loud but can also be used in mixing to tame out peaks in certain frequencies.

Anything that is multiband works best with grouped sounds instead of individual channels (as mentioned many times above).

A limiter will come in handy if the compressor is failing to give you a good, constant volume throughout the arrangement.

It can also be used on sounds that need to be upfront in the mix. But be careful, because this can make things sound overly compressed and unnatural.

UNDERSTANDING EXPANDERS & GATES

These two dynamic tools are used to remove unwanted noise from a sound source.

Unwanted noise includes tape hiss, mic bleed, ambient noise, and background noise.

While compressors can compress both upward and downward, these can only compress upward.

However, in practice, we typically encounter them as downward expanders, and in this format, any signal falling below the threshold is reduced in level.

Expanders typically use a ratio of 2:1 to 3:1 to reduce unwanted noise.

Gates, on the other hand, do not rely on ratio; they simply remove all sound.

If you know how to use a compressor, then you know how to use expanders and gates because they have similar (if not the same) parameters.

Gates can also be used in a creative way to add great effects to any sound. They're also used on reverbs to create what is known as "gated reverb."

The hold and release parameters are used in gated reverb to tailor the reverb tail so that it drops abruptly after an initial decay

phase. I don't use this technique at all; I've never had a reason to use it.

Every time I use a tool, I use it for a reason, and I've never come across a problem that needs a gated reverb.

An expander is mostly used when the gate is adding click noises to your recording or if the gate just acts too rapidly on a sound.

Expanders will definitely be a good choice in that case.

There's not much to discuss about expanders and gates because they're typically used for removing mic bleed.

These days, we have great gate dynamic tools that have a look-ahead feature, so expanders are used less these days.

TRANSIENT PROCESSORS

The transient shaper is not an old tool like other effects and processing tools. It's not widely known in the game, but it's now widely used and a fantastic tool for adding punch to a sound.

It usually has two parameters, which are attack and sustain. It is used to increase or decrease the attack and sustain of any sound.

The transient shaper is mostly used on percussive sounds, such as drums. It doesn't sound great when used on sounds that have a long sustain or release time.

Basically, it's a processor that tracks and reshapes the envelope of an audio signal.

Unlike other dynamic tools, the transient shaper is the only processor that doesn't react directly to level. There is no threshold to deal with here.

It is used to make shorter sounds a bit longer or to make a long sound short and punchy.

You can change a long and boomy kick drum sound into a tight and punchy kick.

It is easier to get carried away with this tool and end up with a distorted mix.

So use it on group channels (especially the drum bus) and always A/B test your results.

Also, don't be fooled by the increased volume it will give your sounds.

Do level matching with the dry signal and the affected signal when you're doing the A/B test.

There are no rules when it comes to transient shapers, or any suggestions for that matter.

Simply explore and see if it helps improve your sounds.

Using EQ Like A Pro

This is the most difficult part for anyone who is new to music mixing. It can be really hard to know when and why you need to use an EQ (equalizer).

The number-one rule is to never fix something if it's not broken.

You need to know why you're applying an EQ and what the outcome is going to be before you even put it in your inserts.

Do you want to fix mud, remove harshness, solve nasal noise problems, or make space for another instrument?

Those are the types of questions you should ask yourself.

WHAT IS EQUALIZATION?

EQ = Short For Equalizer

EQ is basically the process of adjusting the levels of particular frequencies. It is a more detailed level control that allows an engineer to cut or boost levels at specific frequencies.

You can shape individual sounds in a mix to make them fit and work well together as a whole. With EQ, you are able to add personality and character to sounds in a mix.

I won't get into the background and history of an EQ in this guide, I'm only going to focus on how you can use an EQ more effectively during the mixing stage.

Most home stereos and DJ equipment use simple filters that allow you to adjust the bass, midrange, and treble.

However, most complex equalizers, such as graphic and parametric, are flexible and can be used to shape the frequency content of an audio signal.

Equalizers can also be used during recording, broadcast, live performance, mastering, and mixing.

An EQ can be used to remove unwanted sounds, enhance specific tones of a sound, remove feedback, or maybe just to make a voice more prominent.

The parametric and graphic equalizers are the most commonly used equalizers, particularly in music creation and engineering.

Graphic equalizers are often included in consumer audio equipment and software like Windows Media Player, a home and car stereo sound system, or a DVD player.

A parametric equalizer requires much more expertise, and it is really flexible, which makes it really great for audio engineering.

First, we'll look at different types of filters before we get into the parameters.

FILTER TYPES

High-pass and Low-pass Filters

The first one is the high-pass filter, which is also known as a low-cut filter. in

This one cuts everything below a certain frequency and leaves anything above that frequency unaffected. The opposite of this is the low-pass filter, which is also known as a high-cut filter.

This one removes everything below a certain frequency. These 2 filters are also called "one-sided" filters because they can only cut frequencies; they can't boost.

They also have a slope setting to determine how steep the cut will be.

The slope settings are configured in multiples of 6db per octave, such as 6db, 12db, 18db, 24db, 48db etc.

Shelving Filter

Then you have the shelving filters, which are almost the same as the low- and high-pass filters except that the shelves can cut and boost as well.

The low- and high-pass filters are used to completely remove frequencies, but shelving is used to get more tone control.

Instead of removing all frequencies at a set frequency, they reduce or increase signals above or below the set frequency.

These are also "one-sided" filters, and they don't use a slope; instead, they use a bandwidth to control how steep the cut or boost must be.

Bell Filter

This one is called a "bell filter" simply because it is shaped like one, and it is the most commonly used EQ type.

This one allows you to use all independent EQ controls, which are gain, center frequency, and bandwidth.

This type of filter is used for more accurate amplification adjustments.

It allows you to boost or cut specific frequencies.

These three filters are the most frequently used in audio mixing.

UNDERSTANDING EQ PARAMETERS

The EQ parameters are really easy to understand as compared to other audio processing tools.

Now that you know the most commonly used filters in audio mixing, let's look at the EQ parameters.

Gain: This parameter is used to control the amount of a cut or boost that you want to create.

Frequency: The center frequency is used to determine where you want the EQ band or filter to be placed in the frequency spectrum.

Bandwidth: This parameter controls the range of frequencies you want to affect, determining how wide or narrow the cut or boost must be. Some equalizers call this the "Q" because the word "bandwidth" is too long and can't fit in the console.

DIFFERENT TYPES OF EQUALIZERS

There are 3 different types of EQs that are used in the studio: graphics EQ, parametric EQ, and roll-off EQ. The rollofs are simply your high-pass and low-pass filters.

The high-pass filter allows you to cut out lower frequencies and leave the high frequencies unaffected.

The low-pass is the opposite, which allows you to roll off the high frequencies and leave the low frequencies unaffected.

A graphic EQ has a volume control for each frequency. You can turn each frequency up or down using the volume slider.

A parametric EQ is much more complex as compared to a graphic EQ.

A parametric EQ can control three aspects of each frequency using three different parameters: gain, center frequency, and Q-factor (bandwidth).

The major difference is that a graphic EQ has fixed frequency controls, while a parametric EQ is flexible.

For instance, a graphic EQ can have a volume control at 40Hz, you won't be able to change that.

But with a parametric EQ, you can simply move the frequency control to 43Hz if that's what you want, which gives you more flexibility.

A parametric EQ also controls the Q-factor or bandwidth. This helps you control the range, which will determine how wide or narrow you want the cut or boost to be.

A parametric equalizer is always a good choice for audio engineering because it allows much more detailed control of a sound and allows you to make finer adjustments.

UNDERSTANDING FREQUENCY RANGE

It is very important to know different frequency ranges by name.

If you didn't know, it's in our DNA to perceive sound and know all the frequencies from 20Hz to 20kHz by heart.

Every cell in our body is designed to know frequencies.

Even NASA found sound frequencies in space, but they couldn't figure out what sound has to do with the universe.

You know why? Because they don't read the Bible; believe me, the answer is in the Bible, but I won't get religious on you.

If you can get access to a science lab, then go there and ask the scientists to show you how cells react to different sounds, and you'll understand why some sounds create emotion while others make you feel like dancing.

You'll be able to see that our entire body perceives sound, not just our ears.

Therefore, we all know each and every frequency and pitch by heart; it's in our DNA.

The trick here is to learn them by their names and know what cutting or boosting frequency does to a specific sound.

There are two great tools that I recommend that will help you solve this issue with ease. These are some really good ear training tools, and they're absolutely free.

Feel free to use paid options, but the ones below will help a lot.

1. EQ Match Training

<http://www.easyeartraining.com/topic/audio-mixing>

2. Frequency Quiz

<http://www.puremix.net/ear-training.html>

Different frequencies affect us both physically and psychologically. Not to mention the spiritual aspect.

But the most important aspect is how a combination of frequencies makes us feel.

So the art of audio mixing involves using EQ to place different sounds at specific frequencies to capture the perfect or desired emotion.

THE HARMONIC STRUCTURE OF SOUND

Each and every sound is made up of different frequencies and volumes.

The harmonics account for why one instrument may sound different from another; each sound has its own unique timbre.

Each sound will have different harmonics.

When you boost or cut a specific frequency using an EQ, you're actually removing or adding volume harmonics to the sound.

This is why each sound acts on or responds to EQ differently.

While in the mix, we use EQ to make sure all sounds work well with each other without creating overlapping or frequency masking.

Now let's look at the most common uses of EQ during the mixing stage...

THE COMMON USES OF EQ IN MIXING

For an audio engineer, the first common way to use EQ is to use it in solo mode during the recording stage to make sure you capture the best sound possible.

This is mostly based on whether you're going for something that sounds natural or interesting.

Another application of EQ is during the mixing stage to determine how an audio track should sound in relation to other sounds.

EQ can also be used in live stage performances and during audio mastering.

You can also use EQ to come up with something that sounds unique or interesting.

Most of the time, sounds are equalized to help them work well with each other or to make them sound similar to each other or even dissimilar.

For instance, a lead instrument can be equalized so that it becomes more cutting and grabs attention.

An instrument can be given more bass to make the song more danceable or simply to excite the rap listener.

In other cases, you may need to boost specific frequencies in the bass to make it audible through small speakers and earbuds.

UNDERSTANDING THE FREQUENCY SPECTRUM

The frequency spectrum in music production and audio engineering is known as "height" (from subbass to treble).

Height refers to building a healthy frequency dynamic in a mix that has no obvious EQ gaps.

The frequency of a sound is measured in hertz (Hz).

This can be easily understood as the number of cycles per second of any given wave. A 200Hz tone has 200 cycles per second, which ultimately provides us with a recognizable pitch.

The frequency range that our ears can detect is from 20Hz to 20kHz.

That's the range of frequencies our ears are able to hear, and as you grow older or listen to loud music, the range decreases.

Humans can't hear anything above 20kHz, and anything below 20Hz is felt rather than heard.

Before we begin EQing, we should understand the frequencies we are about to adjust.

It's important for a mix to have the following frequencies in appropriate and well-balanced percentages:

Low End (20Hz - 160Hz): This is where you'll find your sub and bass instruments. This range of frequencies needs to be treated with care, and most of these frequencies are generally felt rather than heard.

Low Midrange (160Hz - 500 Hz): This is the most difficult frequency range to master, and it is the foundation of all lower range instruments. Boosting these frequencies can make a sound thick, or even worse, make the sound muddy and boxy.

Mid-Range (500Hz - 2.5kHz): This is where you find a lot of frequency overlapping, as most instruments sit somewhere in this range. There's also a lot of nasal quality, so you'll be spending more time in this range.

High Mid-Range (2.5kHz - 8kHz): This frequency range has a lot of upper harmonics and sound clarity. This is where you'll find the attack of a snare drum or the pluck of a guitar for example.

High End (8kHz - 20kHz): This is where you mostly find the air or sizzle of sounds. These frequencies can cause fatigue if you

add too much energy, or they can make a mix dull if there's too little high-frequency energy.

USING EQ IN A MIX LIKE A PRO

Now that we understand where our frequencies are, we need to learn about frequency spectrum management.

The correct manipulation of equalization is an integral part of achieving a punchy, clear, and full-sounding mix.

EQ can be used to thin out certain sounds by cutting the low and lower mid frequencies, or conversely, to fatten up a sound by boosting the low frequencies.

Boosting mid-range frequencies can also make a sound cutting and edgy, while cutting these frequencies will soften or sweeten the sound.

Any skilled engineer or producer should know when and where to apply the correct EQ in order to achieve a good height (sub-bass to treble) that won't make the final mix lack vital frequencies or have gaps.

The mixing engineer must oversee these aspects to ensure that there is enough punch, presence, or crispness in the mix to keep the listener glued to the stereo from start to finish of the song.

The frequency spectrum (20Hz to 20kHz) needs to be treated with a lot of care without messing up the pitch or timbre of a sound.

This is a great skill that audio engineers need to learn and master.

Think of EQ as the primary application for spectrum management.

But before you can start using an EQ, you need to first understand frequency overlap, or "masking."

This happens when two or more different instruments sound good when boosted at the same frequency.

For instance, you can find instruments that sound good when both are boosted at 2kHz.

If you boost a guitar as well as the vocals at 2kHz, they'll both sound good on their own (when played solo), but when played in the mix, they will overlap and create masking.

So to make them work well together with the rest of the sounds in a mix, you can boost one at 1.5kHz then boost the other at 2.5kHz.

You can also swap the frequencies (1.5kHz and 2.5kHz) around for both instruments to see how they'll work with the rest of the sounds in the mix.

When you're using an EQ during the mixing stage, you will need to figure out what frequencies are available for each instrument and allocate them accordingly.

For instance, you can place an acoustic guitar around 6kHz, the hi-hats at 10kHz, electric bass harmonics at around 900Hz, the attack of the kick drum at 4kHz, the piano at 1.3kHz and finally your vocals at 2.7kHz.

I know what I just mentioned above is not really accurate, but I'm just trying to give you a rough idea about how to allocate sounds in the frequency spectrum in order to get a great-sounding mix and avoid masking or frequency overlapping.

This is what I call spectrum management.

To avoid any confusion, you can mix with a notepad and write down each frequency you boost or cut.

Simply place the notepad somewhere you can easily access it so you can always refer to it and add more notes without being distracted or losing focus.

This will make your EQ workflow a lot faster.

With a lot of practice and some ear training, you'll be able to know which frequencies need to be cut or boosted.

A simple way is to sweep around the frequency spectrum using a bell filter with a 10dB boost and a narrow bandwidth, then just go around the spectrum to find problem frequencies and then remove them. Or boost wherever necessary.

GENERAL EQ FREQUENCIES

40Hz Hz: You can remove these subsonics from your sounds using a shelf or high-pass filter if necessary. These are not audible, and they will give you more headroom when cut.

100Hz to 300Hz: This is the part that mostly gets clutter and may cause your mix to sound muddy, so cut around 250Hz with a wider Q. But be careful; some sounds may be dominant in this space.

1000Hz to 8000Hz: If you boost too much of this part of the frequency spectrum, you can make your mix very tiring. Yet this is the part you need to use to make your sounds present in a mix.

10000Hz to 16000Hz: Adding at this part of the spectrum can make a sound more exciting, while cutting can make it mellow. Most sounds need some air, and this is where you boost, but if you boost everything, your sounds will clash and you'll have masking.

USEFUL FREQUENCIES FOR SEVERAL INSTRUMENTS

Vocals: presence (5kHz), sibilance and clarity (6kHz - 8kHz), boxiness/muddiness (250Hz - 800Hz), fullness/boominess (100Hz - 250Hz)

Electric Guitar: fullness (100Hz - 240Hz), muddiness (250Hz - 800Hz), bite (2.5 kHz), sheen/sizzle (8 kHz)

Bass Guitar: bottom (40 - 80 Hz), attack (700 - 1000 Hz), string noise (2.5 kHz)

Other Bass Sounds: bottom (50Hz - 100Hz), roundness (100Hz - 250Hz), muddiness (250Hz - 800Hz), attack (800Hz - 1kHz)

Snare Drum: fatness (240 Hz), crispness (5 kHz)

Kick Drum: bottom (60 - 80 Hz), muddiness (250Hz - 800Hz), slap (4 kHz), presence & punch (1kHz - 8kHz)

Hi Hat & Cymbals: muddiness (250Hz - 800Hz), sizzle (7.5 - 10 kHz), presence (1kHz - 6kHz)

Toms: attack (5 kHz), fullness (120 - 240 Hz)

Acoustic Guitar: harshness/bite (2 kHz), boominess (120 - 200 Hz), cut (7 - 10 kHz)

Piano: bottom (50Hz- 100Hz), fullness (100Hz - 250Hz), muddiness (250Hz - 1kHz), presence (1kHz - 6kHz), clarity (6kHz - 8kHz), air (8kHz - 12kHz)

Strings: bottom (50Hz- 100Hz), fullness/body (100Hz - 250Hz), muddiness (250Hz - 1kHz), crunch (1kHz - 6kHz), clarity (6kHz - 8kHz), air (8kHz - 12kHz)

This frequency guide will help you see where each sound has its essence or important frequency.

Use this as a reference when you cut or boost any sound so that you know whether you're benefiting the sound or not.

A lot of people make the mistake of cutting out all the muddiness in each sound, which can result in an unnatural or pinched-sounding mix.

Yes, muddiness is an unwanted sound, but you don't want to cut it all out. Removing all mud will make your mix sound unnatural and unbalanced.

Also, be careful not to clutter a certain frequency by boosting a lot of different sounds in the same frequency range.

This is why you want to always use subtractive EQ instead of boosting.

Using subtractive EQ will help you clean out your mix, but feel free to boost if necessary. A mix that only has subtractive EQ sounds weird and thin.

How To Use Dynamic EQ In Mixing

In this section, I'll show you how to use dynamic EQ, which is a technique that is rarely used by most mixing engineers.

It is mostly used in the mastering stage, but today we're going to look at how to use dynamic EQ in mixing.

The major difference between a static and dynamic equalizer is that when you create a cut or boost on a static EQ, it stays constant throughout the whole song.

A dynamic EQ only creates the cut or boost when the signal hits a certain threshold. This will keep your mixes punchy.

It is a very handy tool for fixing a specific problem frequency, not a good tool for removing unwanted frequencies like mud, rumble, boxiness, nasal, harshness etc.

Not to say you can't use it to remove unwanted frequencies; you can if you think it'll work well with the material you're working on right now.

If it sounds good, then there's no reason you shouldn't use it.

But it works well for fixing dynamic problems, more like a multiband compressor, which is why it is mostly used in mastering when the engineer can't go back to the mix to fix the problem.

Before dynamic equalizer tools were available, you had to automate the EQ in order to achieve the same effect, which can be really time-consuming.

Sometimes you may find that the cut or boost you create works well in certain parts of the song but doesn't sound good in other parts. That's when dynamic EQ can come in handy as well.

Most of the time, you'll find that using a compressor on things like hi-hats and cymbals doesn't give you the desired results. Instead, you can use dynamic equalization to solve the problem, even for things like overheads.

It can also be used to fix harsh vocals, whereby if you create a big cut to fix the harshness, the vocals start to sound dull in other parts of the song when you're using a static EQ.

It is also great for fixing sibilance and plosives sounds; the letter "P" can create a lot of nuisance, which can be easily fixed with dynamic equalization.

Some singers sound good when they sing soft parts but create a lot of piercing nasal sounds when singing high notes, so instead of using a static EQ, which will make the soft parts dull, you simply fix the problem frequencies using a dynamic EQ.

Instruments can also sound harsh, especially a guitar solo or solos in general.

Some notes can resonate and have a lot of sonic flaws; other notes will play louder than others.

And when you bring up the volume, some notes may sound too loud; that's when you can use dynamic equalization to fix the problem without ruining the sound.

You can also do some mid-side processing and sidechain with dynamic EQ; the possibilities are just endless.

Another great feature is that you can adjust the attack and release times just like you do on a compressor.

Sometimes a compressor, especially if it's working too hard, can kill a lot of harmonics, but using both dynamic EQ and a compressor can result in a really smooth sound.

I find that dynamic EQ works really well to boost or add things like presence or brightness because the boost is controlled dynamically.

That way, you end up with a really pleasing tonal balance.

I wouldn't recommend anyone use it as their go-to plugin for equalizing, but only to **fix a specific problem frequency** that occurs in certain parts of an arrangement or song, especially if your music is recorded live.

USING EQ MATCH TO IMPROVE YOUR MIX

Some EQ plugins these days come with the function of EQ matching. The most popular ones are the FabFilter Pro-Q and the iZotope EQ.

Getting the right EQ settings can be a challenge for beginners, even when they have reference material. It's not an easy task to fix frequency problems in a mix with an untrained ear.

That's where neat tricks like EQ matching come in handy.

Let's say you've done everything you can to get your mix to sound like your reference material, but you can still hear some

faults in your mix, and you have no idea how to fix them because you've tried everything that you can think of.

In this case, it's wise to EQ-match your song with your reference material.

However, make sure that you **do this at the very end of your mix for maximum results**. Doing it at the beginning won't give you great results as your mix is not yet polished.

It will also be difficult for you to determine if you're benefiting your mix or just ruining it.

When your mix is clean and has fewer problems, it will be obvious if you're improving the mix or not, even if you're a complete beginner.

Since this trick is plugin-based, you'll have to get a tutorial for the specific plugin that you'll be using for EQ matches.

Remember, you don't have to go drastic on this. Just enough to get your mix sounding as close to the reference material as possible; it doesn't have to match the reference exactly.

Each and every song is different, so trying to make your song sound exactly like the reference will make it sound weird and out of balance.

Also, make sure that the reference is in the same key as your song, even though I've already mentioned that it is very important so that you get the best results.

Use the highly recommended plugin REFERENCE to see if you've improved or degraded your mix.

There are no rules or guidelines for this technique; you are going to have to rely on your ears.

FAIR & ACCURATE COMPARISON

I've also mentioned this above, but it's very important.

When you EQ or add any processing tool, check if there are any volume changes.

If there are any volume changes, make sure to adjust the volume so that the affected signal matches the original signal in volume.

A lot of people overlook this and wonder why they don't get pro results.

If the affected signal is louder than the unaffected signal, it's very tempting to think that you've improved the sound. So to avoid being fooled by loudness, make sure that you A/B test the before and after results at the same level as possible.

When EQing a signal, be aware that you're also changing that signal's overall level. Level-matching and an accurate A/B comparison will help you judge how successful your tonal changes are.

The majority of EQ plugins feature an output gain parameter; use it to your advantage and never ever ignore it.

Some EQ plugins also come with "auto makeup gain." When this is enabled, the EQ will adjust its output level in relation to any boosts or cuts that you've made.

That will make it much easier to judge the tonal effects of your adjustments.

Again, this applies to all processing tools.

The EQ is the last processing tool I'm going to cover in this section (how to use effects and processing tools like a pro).

I know I missed a few effects; it's simply because I don't use them and have never seen any Pro Engineer use them. These include modulation tools as well.

Chorus effects are great for creating space and movement but can push a sound back in a mix as well.

They work really well with backing vocal harmonies and an acoustic or electric guitar. It's an excellent ear candy effect that can double sounds.

These are hardly used in modern music because they muddy up the mix.

Phaser and Flanger sound great when used as send effects and introduced in certain parts of an arrangement. This gives drama to your song; it's an old trick but rarely used these days, and it can be excellent ear candy too.

These are genre-based, and since this is a concise guide, I won't cover them.

ACHIEVING A PRO STEREO IMAGE

In this module, you will learn the best techniques for manipulating the stereo image to achieve a richer listening experience.

Unlike back in the days of vinyl, where there were limitations as to what you could do with the stereo field, these days music is released digitally and people use earbuds, so you can do a lot with stereo image shaping.

This is also the most underrated technique, but it could be the missing puzzle in your mixes, and it's tricky to get right.

But by the time you finish this module, you should be able to exploit this technique to your benefit.

Recording in stereo is not enough to make your music stand out. You need to know stereo enhancement techniques to achieve that room-filling, huge pro sound.

Panning

We'll start with panning, which is the most basic tool for stereo enhancement.

Panning allows you to adjust the volume of a sound in the left or right channel and shifts a sound's perceived phantom.

That simply means you'll be creating differences between the left and right speakers. That's a really basic way to produce a good stereo effect.

Super-wide effects can be achieved by panning hard left or right in the stereo image.

Similar to the classic double tracking technique, in which a vocalist records the same part twice, one hard left and the other hard right.

That will create a small difference in timing and phase, which will result in one huge sounding vocal that will work well in the chorus parts.

It helps the listener differentiate the chorus from the verse.

You can achieve a lot with simple panning and, most importantly, make space for sounds in a mix.

Audio Panning Tips

- Make sure that you pan individual sounds/channels don't pan a group channel, you'll end up with an imbalanced stereo field.
- Maintain a good stereo image balance. For instance, if you have a tambourine and hi-hat, pan one left and the other right, which will give you a good stereo balance.

- It's wise to pan sounds that occupy the same frequencies opposite each other because that will increase the presence of each sound.
- Keep the bass and kick in the centre, don't pan them. Panning low-end instruments will give you phase issues and your mix will lack punch.
- Any instrument that is driving the song should be centred like the snare drum, piano and lead vocals.
- Background vocals can be panned to create a wide stereo effect.

But ultimately, no one can tell you how to pan because it's more of an artistic technique. Using reference material is another great way to know which sounds should be panned left or right.

Another neat trick is auto-panning. This one is plugin-dependent; you'll have to find a plugin that has an auto-pan feature to achieve this.

You can also achieve similar results by manually automating the left and right parameters in your DAW. You can draw the automation, but this process is very time-consuming it's better to use a plugin that has an auto-pan feature.

Auto-pan can sound awesome when done in certain parts of the arrangement or when switching from one part of the song to another.

Using Delay To Enhance The Stereo Image

Another good technique to play around with the width and depth of a mix is to use a Ping-Pong delay effect. Sync your delay with the tempo of the song to get a more musical effect.

Mix the original signal with the affected signal; you don't want to put the delay at 100% wet. 25% or less will give you a good effect and enhance the stereo width.

Each and every hit or note has to be clear without being masked by the delay repeats.

You can also achieve what is known as the Haas effect with a ping-pong delay set at 100% wet. You'll have to play with the delay time in this case.

For the Haas effect, the feedback parameter has to be set at 0, the delay time at 30 ms or less, and then add a stereo-enhancing plugin to increase the width of the delay effect.

Play around with the delay time till you get a good stereo width.

These days, most effects come with a stereo width parameter. Use it to your advantage to make your affected signals wide in the mix.

If you want the original signal to be affected as well, then use the effect as an insert instead of using an FX channel.

Phase Inversion

Another trick is to invert the phase of a sound on one side (left or right) and leave the other unaffected.

This is one of the oldest tricks. Each DAW has a phase invert switch (open the manual for your DAW, no excuses).

To achieve this effect, duplicate your sound and pan the original hard left and the duplicate hard right.

This is still a mono signal, but when you click the phase invert switch on one of those channels, you'll get a great-sounding super-wide stereo effect.

The sound will be perceived as wide to your ears because you flipped the phase horizontally by clicking the phase invert switch or parameter.

Dual Mono Effect

This one is similar to the phase inversion trick, but the only difference is that you won't be inverting the phase.

You can achieve the dual mono effect by adding reverb on one channel instead of flipping the phase.

Don't add too much reverb, though; use a small room reverb just to make one side slightly different from the other; it doesn't have to be drastic.

Leave the other channel unaffected. Using a vibrato plugin on one channel can also result in a great stereo effect.

Mid Side Processing During Mixing

In this module, I'll be covering mid- and side-processing.

I'll also share some creative ways you can use it to get the most out of this processing technique.

Some people think that mid-side processing is only necessary in the mastering stage, but it can really come in handy in the audio mixing process as well.

This technique can be used in a lot of different ways, but first let's understand what it is.

What is Mid-Side Processing?

Tools that control dynamics and tone, such as reverb, equalizer, compressor, and modulation effects, will affect either the whole mono or stereo signal of a sound.

However, audio tools that have mid-side processing functionality can affect the mid channel as well as the side channels separately.

The mid channel is the center of the stereo image, and the side channel is the width of the stereo image (left and right channels).

With mid-side processing, you can achieve a full and wide-sounding mix.

The same technique can also be achieved by using other stereo imaging techniques or even panning.

The whole concept of mid/side processing was used back in the 1930s, before stereo playback existed, as a recording technique to enhance space.

But today, this technique is used in a lot of different ways and for many reasons.

Mid- and side-processing when done wrong, can cause phase shift issues or even stereo image imbalance. So use it, but remember that less is more.

Using Mid/Side Processing During Mixing

The most common use of mid-side processing during mixing is to keep low-end sounds such as kick and bass in the center of the stereo field.

This is to make sure that they're in mono, and you can achieve that by using mid/side equalization. You can do that by simply cutting the low-end on the "sides" (left and right channels) of the sound.

Don't cut it out completely; you can push it at least up to 110Hz and leave the top end unaffected for the best results.

Another neat trick is to use this processing technique in your send or return signals.

For instance, in your reverb send channel, insert an EQ plugin that has a mid/side feature, then scoop out the low-mids on the mid channel and leave the sides untapped.

This will keep the reverb from muddying up the mid channel of any sound you use it on.

You can also do a low-cut if you want to make sure the reverb doesn't add muddiness in the low-end.

Boosting the high frequencies on the sides will increase the clarity of the reverb, making the sound that is affected brighter.

Some call this technique "mid side reverb." But you can do the same thing for any effect, such as delay, saturation, or even modulation effects.

If you have a stack of guitars, then route them to a group or bus channel. Then make a high-frequency boost on the sides; that will make them wide, helping them sit well in a mix.

You can automate this by increasing the volume of the Sides only in the chorus parts of the song so that the guitar sounds bigger and wider in the chorus as compared to the verse.

The same technique can be used on vocals and many other instruments. Only your imagination can limit you.

If you want to enhance the room sound of your drum overheads, then make a small high-frequency boost on the sides, but don't overdo it; keep it subtle.

A boost in the mid channel of your drum bus or group will enhance the toms, kick, and snare. Helping them become punchier in the mix.

A lot of plugins have the M/S feature; it's not only limited to equalizers, but you can also find it on compressors and even distortion plugins.

Mid-side compression is a really neat trick to achieve more loudness. You can achieve that by compressing the mid-channel and increasing the gain on the sides.

It's pretty simple but really effective.

You can even be more creative and do parallel compression with the M/S technique.

Another neat trick is to add mid-side compression on a reverb send or FX channel by compressing the mid channel more than the sides.

But note that by increasing the sides, you're making the reverb wider.

My favorite three plugins for doing mid-side processing are the Waves Center, Ozone Imager, and Fabfilter Pro-Q.

For M/S compression, I use the Waves Puigchild. But there are a lot of great tools out there, so it doesn't matter if you're using Ableton, Pro Tools, Logic, or whatever.

As you can see above, the possibilities are just endless. Simply put, take some time to play around with these tools so that when you do your mixes, you can solve problems with ease.

Fix Masking

Mid-side processing can also be used to fix frequency clashing. If you have a piano sound that is clashing with the guitar.

You can simply allow the piano to dominate the center and let the guitar shine on the sides.

A simple cut in the mid channel on the piano and a cut on the sides for the guitar will reduce the frequency masking.

This technique can be used for any sounds that are fighting for the same frequencies in a mix. Except for low-end instruments, because you always want your low-end sounds to be in the center.

Multiband Width

I only use this trick in the final stages of the mix.

The Ozone Imager and the REFERENCE plugin are great tools to achieve this.

Simply use your reference material (songs) to compare the stereo image of your mix.

With the REFERENCE plugin, you can see visually if different frequency bands are too narrow or too wide and then use the Ozone Imager to adjust accordingly till you get satisfying results.

Anything that is multiband is used on a group of sounds instead of one sound.

Unless you want to correct a specific frequency issue with a sound.

I use this technique on the mix bus, but feel free to use it on your group channels if you believe it will give you better results.

How To GET BIG BOTTOMS

No not these bottoms 🍑🍑

I'm talking about the low-end of a mix (sub and bass frequencies).

It's time to get into the meat and cut out all the fat.

By now, you have an idea of how to start and finish a mix, but the big question now is: how do you use all the different tools mentioned above to achieve great results?

We'll start by talking about the low end because I think it's the foundation of getting a great mix. Your mix can lack in the mid-range or treble, but if it's got that banging bass, people will love it.

It is also the hardest to mix right, so if you get it right from the start, the rest of the mix tends to be much easier.

Even though people these days listen to music using smart speakers, earbuds, tablets, and smartphones, getting the low-end banging is still very important.

Not only should you aim to impress your listeners, but you should also ensure that your song is well balanced across all frequencies.

With that sorted, let's get started...

Working With Bass

As recommended above, always group all your bass sounds into a group channel and mix them as one. Even if one part is recorded with an amp and the other recorded via a DI device.

Just make sure that the final recording of each bass already sounds good before mixing. This applies to all sounds in your mix.

If you're making digital music and you have layered bass sounds, then send them all to one group channel and process them as one bass sound, and they'll blend well together instead of trying to mix each one of them separately.

If you're layering sounds, then I would recommend that you use a low-shelf or high-shelf filter instead of removing frequencies altogether with a low-cut (high-pass filter).

If you have two bass sounds playing the same thing, they'll build up in the low-end and create muddiness, so it makes sense to use a high-pass filter.

This also applies to the top end as well; feel free to use a low-pass filter when you're layering sounds.

That will help you avoid making your mix sound muddy, unpleasant, and cheap.

It is also important to choose one bass part that will be the most dominant throughout the whole mix, then use the other parts to

support that bass without taking space for other sounds in the mix.

How To Control Sub Bass

The easiest way to take care of the sub-frequencies is to use a multiband compressor.

The multiband compressor is used to control the sub-frequencies that start at 20Hz up to 60Hz.

So you have to set your multiband compressor to compress everything that's below 70Hz and leave the rest of the upper frequencies untouched.

You don't need heavy compression for this; just a small -2dB to -3dB gain reduction should work. For the ratio, I found that 3:1 works in most cases, but you can also try 2:1 and compare the results.

This will help your subbass sit well in a mix and stay controlled.

It should also be much simpler to blend it with the kick drum without using a high-pass filter.

How Much Bass Do I Need

This is a frequently asked question by new mixing engineers. Questions like, "*How do I know if the bass level is right?*"

Those are valid questions, but to avoid any guesswork, you have to rely on reference tracks. Listen to a couple of reference tracks, then choose one song you like and do the comparison till you get it right.

If you can't get your song to sound like the reference tracks, don't make the mistake of thinking that your song sounds better. You'll be disappointed when testing your mix on other sound devices.

So make sure that you get a similar balance to your reference when it comes to the low-end; if you can match it, then that's really great.

Use as many reference tracks as you can to avoid guesswork.

Getting Great Bass on Small Speakers

Devices such as earbuds, laptops, and phones don't produce bass frequencies, but you do hear the bass musical notes.

That's because of the bass harmonics.

If you can't hear the bass on small speakers, then you'll need to add some harmonics. You can add harmonics by using saturation, overdrive, or distortion processing tools.

This is a trade-off, so be cautious because adding too much can result in a muddy low end.

It can also take away bass in the low-end since you'll be adding harmonics in the mids and highs.

Reverb on Bass

It is not a good idea to add reverb to a bass sound. However, there are genres such as techno where they use sub-heavy kicks or tom samples to create bass.

Adding reverb to that sub-heavy kick or tom creates a dope wash and moody bass sound.

You need to be careful, though. Use a lot of pre-delay to let the transients hit first, followed by the reverb.

Another trick you can use is to sidechain the reverb with the bass so that each time the bass hits, the reverb will duck.

Use an EQ to filter out the high frequencies of the reverb, which will create an authentic, moody effect.

Also make sure that the tail of the reverb is not too long; it should cut before the next bass hit or note.

What If You Still Can't Get A Good Bass

If you're still not getting the power and girth that you need, then try the following tips:

- Flip the phase (polarity). This will depend on the software or plugins that you use. Most DAWs do have a phase or polarity switch, turn it on and off to see if it makes a difference.
- Check if there are any low-end filters that you might have created while layering. Try removing the filters to see if you get better results or not.
- If you have a busy mix try cutting other instruments at around 700Hz to 1kHz that should bring up the bass harmonics and add presence to the bass.
- Use parallel distortion or saturation by mixing the dry signal with the saturated signal. You can use Fx or Buss channels to achieve this.
- Use automation whenever the bass gets drowned in the mix.
- Create a high-shelf to give space for other instruments, you don't need anything above 5kHz for the bass. Shelving high frequencies does help the bass shine in some cases.
- Make sure that your bass, or at least the low end, is playing in mono, not stereo.

- Use a bass enhancer plugin; this works all the time, even on kick drums.
- If all fails then simply change the bass sound or re-record.

Working With Kick & Bass

The key to getting a big and strong bottom end is determining which sound between the bass and kick will dominate certain low-end frequencies.

You have to determine which sound will provide the weight at the very bottom end and which one will create the drive in the upper low frequencies.

This also depends on the genre you're currently working in.

You can also use a frequency analyzer to figure out which sound will add the drive and which one will add the weight.

Let's start by discussing the kick drum.

Achieving Great Sounding Kick Drums

A kick drum is usually thought of in two parts, which are the transient and decay. Attack and tail are other terms for the same thing.

The attack (transient) is responsible for the smack and for helping the kick cut through the mix. The decay is the low-frequency

resonance of the sound, which gives you the deep pitch of the drum.

It's much easier to control these when you're using a synth kick or a sample.

When it comes to acoustic drums, you have to rely mainly on the drummer to control the attack and tail throughout the arrangement.

Synth drums and samples can easily be pitched to the key of the song to help them blend well with other sounds in the mix.

With an acoustic drum, you don't have to worry about keys; you just have to focus on the tuning of the drum sound.

KICK DRUM LAYERING TIPS

I know this is not part of mixing, but a lot of people always get this wrong and wonder why they can't get a good sound.

So here are some tips, and these tips will work on other sounds as well:

1. Use 2 or 3 different kick sounds, anything more than that is just overkill.
2. Each kick should serve a different purpose: one for the low end, another one for the midrange, and the last one for the top end or click.

3. Acoustic kick drums always work best for the midrange.
4. Layer kicks that have the same tone, pitch and key. Or simply make them play in the same tone or key.
5. Make sure they have the same or similar decay time.
6. Remove or cut out frequencies you won't need on each kick drum. This is where low-cut and high-cut is acceptable instead of shelf filters.

Now let's look at some processing tips for the kick.

ENHANCING AND PROCESSING AN ACOUSTIC KICK DRUM

When it comes to processing any acoustic instrument, the first thing you have to consider is the microphone bleed.

The kick drum microphone will always pick up the other sounds of the drum kit.

This is also known as a "mic spill." Some jazz mixing engineers will leave the mic bleed, while in other genres it's removed completely.

Removing Mic Bleed

The easiest way to remove mic bleed from your sounds is to use a gate processing tool or plugin.

The FabFilter Pro-G is currently the most popular gate because it comes with a look-ahead feature that helps you keep the transient or attack of the kick drum intact.

Other gate plugins might chop off the attack of the kick drum, and that will result in a weak sound that lacks punch.

If you over-do the gate processing, you might add an unwanted click sound to the kick drum.

If you don't have a gate processing tool that has a look-ahead feature, then leave a little bit of mic bleed to make sure that you don't mess up the attack of the kick drum.

A great way to avoid adding unwanted clicks or messing up the transient of the kick is to use an expander instead of a gate.

Expanders use a low ratio setting and will leave some mic bleed.

COMPRESSION & BASS ENHANCEMENT ON ACOUSTIC KICK DRUMS

Compression is not only used to even out the overall level of the kick throughout the arrangement, but it's also used to add more attitude to the sound and increase its energy.

You can achieve that by using a slow attack time with a fast release time.

Those compressor settings, together with a bass enhancement plugin such as the Waves MaxxBass, UAD Voice of God, or RBass, will add more punch to your kick drum.

The bass enhancer plugin adds more bottom end to your kick and broadens the frequency range of the transient.

EQUALIZING AN ACOUSTIC KICK DRUM

When it comes to equalizing an acoustic kick drum, you need to pick harmonics that resonate with the song and the drum itself.

If your acoustic kick lacks some punch, then a boost around 70Hz will definitely help. I say around because this is based on the key of the song and the tone of the sound.

Simply sweep around that frequency range till you find a sweet spot.

If you can't hear the kick drum on small speakers, headphones, or laptops, then a boost around 100Hz to 120Hz will help the kick be audible on small speakers.

But be careful because the bass is usually dominant in those frequencies as well, so choose the frequency to boost wisely or simply cut those frequencies from your bass sound.

To create more clarity for your acoustic kick, cut around 360Hz to 800Hz, this will remove any muddiness and create more clarity.

To add the click or more attack, you must sweep and create a boost around 2kHz to 4kHz. This will help the kick drum cut through the mix.

If your kick drum has too much attack, simply create a high shelf filter to bring in the higher frequencies.

PROCESSING A **SYNTH** OR **SAMPLE KICK**

The same compression and bass enhancement techniques apply to a VST or sample kick drum. The only difference is in the EQ settings, which slightly differ.

First, make sure that your synth or sample kick is tuned to the key of the song, or else it will be hard to make the kick fit well in the mix.

EQ Settings:

If you have a boomy kick drum, then create a low-shelf at 20Hz this will create space for your bass and add more headroom to the overall mix.

If the kick lacks punch, then a small boost around 96Hz will add the necessary punch and fill the space left by the bass.

The bass enhancement will most surely add some mud to your kick, and to fix that, you'll need a cut around 200Hz. That will add clarity to your sample kick drum and remove any boxiness.

To add more snap and attack, make a small boost with a wide Q (bandwidth) at around 2kHz to 4kHz.

A transient shaper is another processing tool that works well with kick drums, especially if the kick lacks attack and is too boomy.

So also try adding a transient shaper to see if it improves the sound or not.

Achieving a Solid, Thick & Weighty BASS

The tail of a bass plays a big role when it comes to mixing bass because of how humans perceive the bass sound. Shorter notes will appear quieter to our ears, while longer notes will appear larger.

The bass also follows a similar pattern as the kick drum, with an attack and tail, but the attack of the bass is usually in the midrange but can also be higher on other bass sounds, like the slap bass, for instance.

The tone is determined by the number of harmonics. For example, a harmonically dense bass guitar will have a lot of mids as compared to a sine wave or sub-bass.

It is believed that certain frequencies hit certain parts of our bodies. The ultra-low frequencies are mostly felt rather than heard, and they shake your pants.

The frequency range from 50Hz to 80Hz is known to hit your abdomen, while 80Hz to 100Hz hits your belly, and finally 100Hz to 140Hz hits your chest.

So it's very crucial to decide if you want a bass that hits the abdomen, belly, or chest.

If you choose a bass that hits the belly (stomach), then the kick has to hit the chest, or vice versa. Always keep that in mind when mixing low-end frequencies.

How To PROCESS A BASS SOUND

When it comes to processing a bass sound, I recommend that you start with a multi-band compressor to take care of the lower bass and sub frequencies, as I've already mentioned above.

Adding Compression to a Bass

How much compression is needed will really depend on the dynamics of the sound.

However, you don't want to compress a sine wave bass sound since a sine wave is already square and doesn't have dynamics.

Adding compression to a sine wave will make it weak, so instead use the envelope settings to shape the sound.

If you're working with a real bass guitar, then for compression, you use a fast attack time, but make sure that you don't choke the transients.

Try somewhere between 5 ms and 15 ms. That's a good start for the attack.

For the release time, use a fast to medium release time. Avoid a release time that is too fast, or else your bass will distort.

A sweet spot is usually around 50 ms to 200 ms.

Also refer to the compression tutorial above to get the best settings for your bass sounds.

How To EQ a Bass

When it comes to equalizing the bass sound, the most important frequencies range from 60Hz to 110Hz.

If the kick drum is providing the weight and the bass is driving the song, then add a low-shelf filter to around 60Hz to carve out some space for the bass.

You don't need to remove those sub-frequencies completely; simply bring them down a bit. If you remove them, your low end will most likely sound weak.

In case the shelf filter makes your bass sound thin, you can compensate by creating a boost around 80Hz to 110Hz. This will also depend on where you boost your kick drum.

If you boost your kick at 90Hz then don't boost the bass at 90Hz, try 80Hz or 110Hz.

A small cut on the kick where you boost the bass will help create more space for the bass.

The same is true for the kick: if you boost certain frequencies on the kick, you must cut the same frequency on the bass.

But only do this for the bass frequencies (60Hz to 120Hz) ; you don't need to do it for the midrange and high frequencies.

Most professional engineers, including Chris Lord-Alge, never touch a bass sound's sub- and low-bass frequencies.

I mentioned the tips above just in case your kick is clashing with your bass.

However, by all means, try not to cut the low and low-mid frequencies of a bass sound.

Chris Lord-Alge says it messes up some of the bass notes, and I totally agree. So I don't touch any of the lows and low-mids for the bass sound.

To add midrange punch to your bass, create a boost around 700Hz to 1.6kHz.

To know how much boost or cut to apply, simply listen to your reference tracks.

HELPING THE BASS CUT THROUGH THE MIX

To help the bass cut through the mix, there are a few things you can do with saturation, overdrive, and distortion.

If your bass is clean, then duplicate your bass sound and apply an amp simulator plugin to the duplicate track. This will squash the signal and add a different character to it.

Blend the distorted sound with your clean bass sound, but don't make it too obvious.

Remember, *less is more*.

By combining clean and heavily distorted sounds, your bass will be able to cut through the mix and be heard in small speakers and earbuds.

Use parallel saturation to add more warmth and depth to your clean bass.

If the bass is already saturated or recorded through an amp, then you can skip this step and simply blend your DI bass with the amp bass.

Another great tool to use to get a really good and solid bass is to add a bass enhancer plugin, such as the Waves RBass, Waves LoAir, UAD Precision Enhancer, and PSP MixBass2.

Feel free to use any bass enhancer tool that you like, but the ones I mentioned above are the ones the pros use the most.

If you use a subwoofer, you'll be able to hear the sub shaking things, but that might trick you into thinking you have enough but leave you disappointed when doing a car test or playing your mix on small speakers.

So make sure to use reference material to avoid adding too few or too much.

Another good trick is to use parallel saturation. Use an FX or bus channel to blend the dry signal with the saturation.

Add an EQ on the FX or bus channel so that you can cut out any low frequencies that you've boosted on the kick drum. Or else your low end will be muddy and lack clarity.

A cut of about -3dB at 80Hz to 100Hz always does the trick, but always feel free to experiment.

The reason I recommend parallel saturation, overdrive, and distortion is because adding saturation or distortion directly can kill the transient and the dynamics of the bass.

So it becomes much easier to control the amount of saturation and distortion when it's on an FX or bus channel. You get much more flexibility and can also use tools such as EQ to polish the saturation.

Another neat trick is to use a multi-band saturation tool such as FabFilter Saturn or Izotope Trash to only beef up the midrange and leave the sub and high frequencies untouched.

ADDITIONAL TIPS TO HELP THE KICK & BASS BLEND WELL TOGETHER

Here are some extra tips that will help you get a big and phat low end in your mixes.

Complementary Kick + Bass

Choose sounds that complement each other. Don't use a boomy kick with a bottom-heavy bass sound. This will create mud in the low end, and it will be hard to mix.

If you choose a boomy kick drum, then choose a lighter-sounding bass sound.

Choose a lighter-sounding kick drum if you have a sub-heavy bass sound.

Choosing sounds that complement each other makes mixing easier.

Length of The Sound

Some genres use big and heavy basslines. In that case, it's always wise to put your kick drum into a sampler and shorten the length of the kick using an envelope.

This will help you avoid any clashes.

If you have shorter bass notes, then you can try using a long-tail kick drum.

Simply listen to reference tracks to determine the length of each sound and adjust accordingly because each and every genre is different.

Low-Pass Filter

A good way to compare the low end of your tracks with the reference is to use a low-pass filter on both the reference tracks and your mix.

This will help you listen to the low end without the other frequencies interfering.

A plugin such as REFERENCE makes this much easier because you can solo the low end on your reference material with a simple click of the mouse.

Group The Lows

Another good trick that is mostly used on R&B and hip-hop songs is to send both the kick and bass to the group channel and then add a limiter with a gain reduction of about -6dB.

Make sure that your bass and kick don't have any compression before doing this, though. This is the easiest way to blend the two sounds together.

Use Automation on Your EQ

Sometimes you'll notice that you made your bass weighty and heavy, resulting in a lighter kick that works well with the overall mix, but when the bass drops in the arrangement, the kick sounds too thin.

In that case, you're going to have to automate your EQ settings to make sure that your song becomes punchy throughout the entire arrangement of the song.

The same applies if the bass becomes thin when the kick drum is not playing.

This also applies if you're using sidechain in your mix; you "might" want to remove the sidechain when the kick drum is not playing. But this depends on taste and genre.

I won't be covering sidechain in this section since I have a dedicated sidechain chapter in this guide.

I'll share with you some really neat sidechain strategies in the upcoming modules.

6 EASY WAYS TO ELIMINATE PHASE CANCELLATION

Phase cancellation is the silent killer of great mixes.

It destroys both impact and low-end. It either makes tracks sound thin and lifeless or causes them to disappear entirely. But most of all, it's seriously frustrating.

(If you've ever wondered why a kick still sounds wimpy after adding 18 dB at 60 Hz, you know the feeling.)

At its most basic, phase refers to sound waves—simply put, the vibration of air.

When we listen to sound, what we're hearing are changes in air pressure. Just like the ripple of a stone in water, sound is created by the movement of air.

And just as in water, those movements cause a rippling effect—waves comprised of peaks and troughs. Those waves cause our eardrums to vibrate, and our brains translate that information into sounds.

When we record sound, the diaphragms in our microphones essentially replicate the action of our eardrums, vibrating in accordance with those waves.

The waves' peaks cause the mic's diaphragm to move in one direction, while their troughs generate movement in the opposite direction.

Phase cancellation occurs when two signals of the same frequency are out of phase with each other, resulting in a net reduction in the overall level of the combined signal.

If two identical signals are 100% or 180 degrees out of phase, they will completely cancel one another if combined.

When similar complex signals (such as the left and right channel of a stereo music program) are combined, phase cancellation will cause some frequencies to be cut while others may end up being boosted.

Phase and phase difference are real-world issues in areas such as audio equipment electrical wiring, signal path, and microphone placement during the recording process.

Phase reversal can be a serious compromise of sound quality or a special effect affecting the perceived spaciousness of the sound, depending on the context of its occurrence.

I'll share a few tactics you probably haven't heard before. Implement them to add punch and weight to your mixes.

1. Fix Phase Cancellation From The Beginning

The best time to fix phase cancellation is at the beginning of a mix. There's no use adding mountains of EQ or compression if a simple polarity flip can give you what you're looking for.

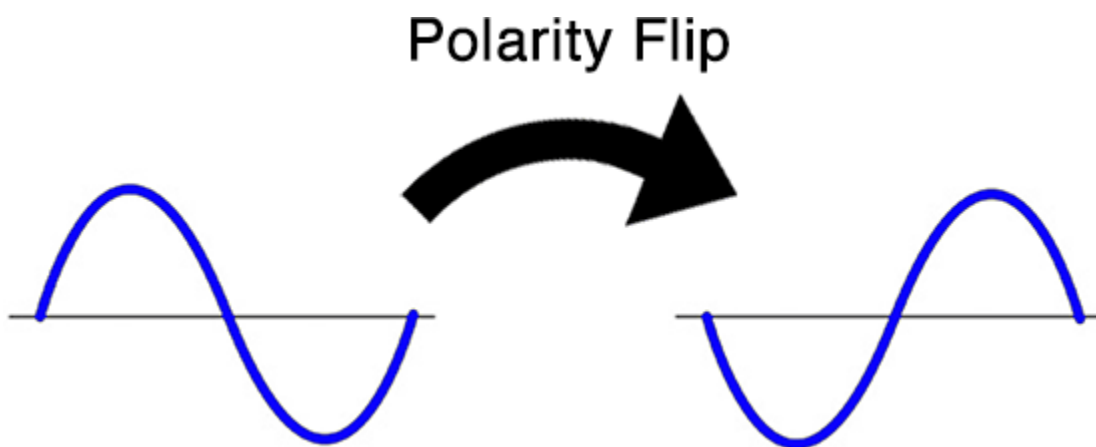
Check for phase cancellation during mix prep before you add any processing. Make this a permanent part of your workflow.

Remember: phase cancellation can crop up in any mix, regardless of genre. Finding and fixing it early on will make mixing faster, easier, and more fun.

2. Go Beyond Polarity

When checking for phase cancellation, flipping the polarity is the first step. But this is just the beginning.

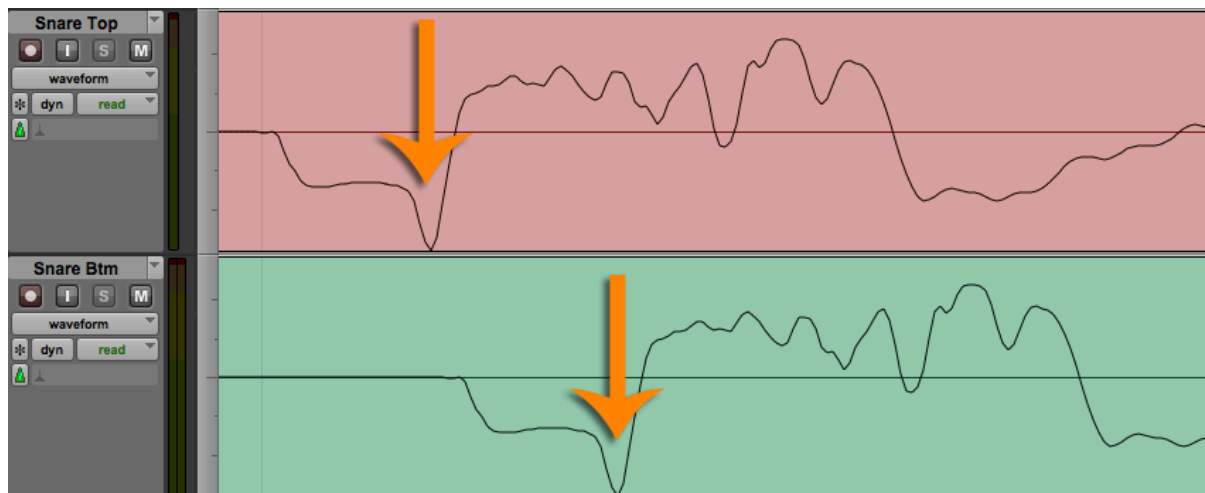
First, realize there's a difference between polarity and phase. When you flip the polarity on a track, you're actually just reversing the direction of its waveform.



While this can help, it often won't fix phase cancellation problems completely.

If two tracks are only slightly out of phase, for example, a polarity flip alone will leave you stuck choosing between the lesser of two evils. For this reason, it's worth exploring other options.

For starters, zoom in on the waveforms in your DAW and compare the peaks and valleys of different tracks. Here's an example of two tracks that are slightly out of phase:



When dealing with the above, nudging one track forward or backward in time often works better than flipping the polarity.

This approach works well when you have two mics that were placed at unequal distances from an instrument (e.g., inside and outside kick drum mics).

For an easier solution, try [Sound Radix' Auto-Align](#). This plugin will automatically optimize timing relationships between different tracks.

With plugins like these, it's tempting to go trigger-happy and optimize everything.

There is a trade-off, however.

The natural time delay between different tracks often contributes to an instrument's depth.

While time alignment can yield a tighter, punchier sound, it can also flatten and destroy this depth.

Listen for depth when making time-alignment decisions. While certain genres may benefit from a punchy, one-dimensional sound, others may warrant a more natural approach.

Not all phase "problems" need to be fixed (see #6 below).

3. Check Layered Drum Samples

Many mixers forget to check for phase cancellation between layered drum samples. Follow the steps below when you have multiple samples that play together:

1. Pick a "master" drum sample that you'll compare the others to. Solo it.

2. Solo the next drum sample as well. Listen to the two together.
3. Flip the polarity on the second sample.
4. Listen to the punch and low end of the combined sound. Flip the polarity in and out, and choose what that sounds best. (Sometimes, there won't be a clear answer. Make a decision and move on.)
5. Add in another sample, and repeat steps 3 and 4 until all samples are playing together.

Some mixers take this process further with plugins like [Waves' InPhase](#).

4. Pay Attention When EQing Correlated Sounds

Conventional, minimum-phase EQ alters the phase of sounds. Normally, this isn't a problem. But EQing correlated sounds (like a snare drum that's part of a multi-mic'd drum kit) can cause unexpected problems.

To sidestep these problems, avoid the solo button when EQing correlated sounds. You can also opt to use a linear-phase EQ instead.

Since they don't create phase shifts, they aren't subject to phase-related problems. However, linear-phase EQs are not a one-size-fits-all solution because they can cause other issues, such as pre-ringing.

The best solution is to always consider context. Don't make mixing decisions in isolation. Pay attention to how your decisions affect groups of tracks as well as the mix as a whole.

And avoid the solo button, please!

5. Use Stereo Imaging Plugins With Caution

Obsessing over stereo width is a waste of time.

Why? Because most listeners will never hear it.

Ever walked into a typical living room? The left and right speakers are often in different rooms entirely. Step back several feet from any set of speakers, and you're effectively listening in mono.

And sure, on headphones, stereo width matters. Until someone gives one of their earbuds to a friend...

Many listeners will never hear your carefully crafted stereo image. This doesn't mean you shouldn't focus on it. Just don't get too caught up here.

For this reason, I'm not a big fan of stereo imaging plugins. If you're going to use them, be careful because many of them use phase shift to create width.

The effect can sound great in stereo, but horrible when summed to mono (or even when your mix is played on speakers that are close together).



I don't recommend using these plugins across your whole mix. They can be nice on individual tracks, but often create more problems than they solve.

When using them, check your mix in mono to make sure tracks don't disappear.

6. Use Phase "Problems" To Your Advantage

Not all phase problems should be fixed. Some actually enhance your mix.

For example, phase cancellation between multiple mics on a guitar cabinet can be used to add color and texture to the instrument's tone.

By adjusting the blend between the two mics, you can create a unique sound that cuts through a busy mix.



Phase cancellation can also create depth.

For example, complex phase relationships between mics on a drum kit can smear transients.

This can help the kit blend better and sink back into the mix, instead of sitting right on the front plane of the speakers.

When it comes to phase, a blanket fix-all approach isn't always appropriate.

Always A/B your decisions and listen closely for downsides—fixing one problem can often create others. Most mixing decisions are trade-offs, and the best mixers know when to leave tracks alone.

Article Credits:

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MIXING VOCALS LIKE A PRO

Before we get started, I have to make it clear that this tutorial will work with whatever DAW you're using.

You could be mixing vocals in Pro Tools, FL Studio, Garageband, Cubase, Studio One or using any plugins, whether from Waves, UAD, or stock plugins.

This is a concise guide, so it doesn't matter what you're using, and if you have problems applying the knowledge, then spend some time learning your software or plugins.

I want to give you something that is more detailed and helps you learn how to make vocals stand out in a mix.

That means this module is going to be very long, so grab a cup of coffee, tea, or whatever you prefer, and be comfortable.

The key to getting the right vocal mix is to make sure that you get it right from the source. What you need to understand is that if you put garbage in, garbage will come out.

So make sure that your vocal recording is proper. Once you're sure that you have a good recording that is ready for mixing, then you've won half the battle.

Now the first thing I do is listen to the vocals and try to get a rough picture of the final results in my mind. Once that's done, the next thing is to put all your thoughts into practice.

Tip: There's no right or wrong way for producing great audio mixes; use the rules as a guide and always trust your instincts.

Gate Processing Tips

A gate processor mutes signals with low volume, and it only modifies the RMS level of a signal. Most gates mute the signal completely, while some allow you to set the compression to be just partial.

The use of gates during vocal mixing is to eliminate background noise in parts where the vocalist doesn't sing or rap.

Most controls found on a gate are similar to those on a normal compressor, such as the threshold, attack, and release, and some have a range as well as a ratio parameter.

A gate with most controls will help make this dynamic processing a lot smoother as compared to a simple gate that mutes the sound completely.

For this to work well, I use an expander instead of a noise gate because it is more gradual and it makes it easier to get the right envelope settings (attack and release).

The attack time needs to be fast because a vocal can have parts that are percussive, and if the attack is slow, then the noise gate will open late.

In that case, the first letter of a phrase won't be audible, which will reduce the clarity of the vocal.

Another thing is to make sure that the attack is not so fast that it ends up causing a click sound.

So it is always better to use an expander if you can't get the gate to work or simply use a gate plugin that has a look-ahead feature.

To get a good threshold, find the part where the vocals are really low in volume.

Once you find that lowest point, push the threshold until you start to hear the gate compressing the signal, then push it back up until everything fades in and out smoothly (naturally).

The main aim is to let the gate open when the vocals come in and close when there are no vocals playing.

The tricky part is getting the right threshold and release. So if you're having a problem with the threshold setting, play around with the release.

Find a part where the vocalist sings the longest note and make sure that you hear the whole note before the gate closes again.

If the vocal note cuts, then keep increasing the release slowly till you hear the whole note fading out smoothly. Check that the settings work well with the rest of the vocal.

If you're using an expander and it has a ratio setting, then just know that anything above the ratio of 3:1 is no longer an expander; it's now a noise gate.

Play around with ratio settings below 3:1 to keep things organic and smooth.

This procedure can also be performed manually by zooming in on the wav-form and removing all of the parts where the vocalist is not singing.

This can be a lot of work, but do this if you don't know how to use a gate or expander.

This can result in a lot of click sounds; to avoid that, create a small fade in and fade out to smooth out your manual cuts on the waveform.

You have to do the fades on every cut to avoid clicks.

Equalizing The Vocal

Once you have the background noise cleaned out, you're going to equalize the vocals to remove unwanted frequencies that may clash with other sounds in the mix.

You can use a frequency analyzer to see which frequencies the singer is dominating.

Equalize the vocal while the whole music is playing so that you can hear whether what you're doing is benefiting the mix or not.

Don't solo the vocals when you're adding any processing; the listener won't hear the vocals in solo, so it doesn't matter how they sound in solo just as long as they're working well with the entire mix.

You must use subtractive EQ first and boost after compression for the best results. By the way, you can choose not to follow this rule, especially if you're not using drastic EQ settings.

A parametric EQ is a good choice because parametric EQs are really transparent and allow you to create big boosts and cuts without messing up the timbre.

Vocal EQ Settings

If your vocals sound too thin, find the fullness of the vocal around 100Hz to 250Hz and keep in mind that this frequency range is good for cutting in some cases.

This will add the thickness and warmth that you desire.

Another part you need to cut is the muddiness area, which can be found around 250Hz to 700Hz.

If your vocal has a honky or nasal sound, a cut around 800Hz to 1kHz will do the trick.

If the vocals are harsh, then create a cut with a narrow Q-factor somewhere from 2.5KHz to around 4KHz.

To add more clarity and presence, then do a sweep from 4kHz to 9kHz till you find the right spot, which is usually around 6kHz in most cases. Then boost that using a wide Q-factor (bandwidth).

A narrow cut at around 1kHz to 1.8kHz can add some smoothness to the vocal. This also makes room for other sounds such as guitar and bass.

To add some sparkle and air, a high-shelf boost at around 10kHz will always do the trick.

Keep in mind that these EQ settings are just a guideline; you'll have to sweep around the spectrum to find the right frequencies, but at least with these tips, you'll know where to look.

Use the frequency guide below if you get stuck:

Fullness (100Hz – 250Hz)

Boominess and Muddiness (250Hz – 800Hz)

Honky and Nasal (600Hz – 1.6kHz)

Presence (5kHz – 8kHz)

Sibilance (1.5kHz – 7kHz)

Clarity (5kHz – 9kHz)

Air or Breath (10kHz – 20kHz)

One last thing that's worth mentioning is that mud is not always bad on a vocal, and it's not always necessary to remove the low-mid range when mixing vocals because in some cases it may take away the presence or make your vocals sound thin.

Compression Tips

Before you can start adding your favorite compressor, ask yourself whether compression is needed on the vocals or not; is there a big difference in dynamic range (between the loud and soft parts), and can you fix that manually?

Sometimes you may find that there's a big difference between the loud and soft parts of the recording.

Maybe the vocalist was moving away from the microphone during recording.

You may find that some words are hard to hear or are mumbled.

If that's the case, then the compressor will just ruin the vocal performance.

Manual gain is your best option. You must manually adjust those soft parts to merely match the loud parts.

You can either use volume automation, but I prefer using the Waves Vocal Rider plugin.

It will be easier for you to get the best possible compressor setting that will help the vocals sit well in the mix if the dynamic range is not too large.

Compressors work really well if they're not used as a fixing tool, but instead to polish a good vocal recording or any sound.

So how do you know when to compress?

There are a lot of reasons to compress, but in most cases, you'll find that your vocals are loud in some parts of the mix and sound quiet in other parts.

That's when you'll need a compressor to even out the volume and keep it constant throughout the whole song.

If you're reading this, then I assume you already know what each parameter on the compressor does, and if not, then take a moment to read the dynamics guide above.

Vocal Compression Settings

When it comes to vocal compression, **you need to be armed with the best dynamic processing tools.**

This means you'll need a good compressor with a distinct sound.

Not all compressors are built the same, and they have different noise and tonal characteristics, so you need to find one that will work well with the vocal performance you're currently working on.

Unfortunately, there's no "one size fits all" setting when it comes to vocal compression or compression as a whole. You just need to be hands-on and practice until you get it right.

The music I make is jazz and more African-influenced, so I always keep things as organic as possible, but for some hard-core stuff, feel free to use as much compression as needed.

But too much compression can kill all the dynamics and make the vocal performance sound unnatural.

It will also bring up the background plosives, and sibilance will be exaggerated as well.

The key is to tame out the loud peaks and leave some parts uncompressed.

Step by Step Vocal Compression Process

Compressing a vocal track is a common technique used in audio engineering to even out the dynamic range of a singer's performance and bring the volume of the vocal up in relation to the rest of the mix.

It can also be used to add punch and clarity to the vocal, as well as shape and control the tone.

In this module, we'll walk you through the steps of how to compress a vocal track, including setting up the compressor, adjusting the threshold, ratio, attack and release times, and applying makeup gain.

Before we get started, it's important to note that there is no one-size-fits-all approach to compressing vocals, as every voice and recording is different.

The settings that work for one vocal may not work for another, so it's important to use your ears and make adjustments as needed.

With that said, let's dive in!

Step 1: Set Up the Compressor

The first step in compressing a vocal track is to insert a compressor plugin on the channel strip of the vocal track in your digital audio workstation (DAW).

There are many compressor plugins available, both free and paid, and each one has its own unique set of features and controls. Some popular compressor plugins for vocals include the LA-2A, the 1176, and the FabFilter Pro-C2.

Once you've inserted the compressor plugin, you'll need to set the compressor's input and output levels.

The input level controls how much of the signal is sent to the compressor, while the output level controls how much of the compressed signal is sent back to the track.

To set the input level, start by playing back the vocal track and listening for any areas where the vocalist is straining or pushing their voice.

These areas are likely to be the loudest parts of the track and will require the most compression.

Once you've identified these areas, adjust the input level of the compressor so that the loudest parts of the track are hitting around -6 dB on the compressor's gain reduction meter.

This will give you a good starting point to work from.

Step 2: Adjust the Threshold

The threshold of a compressor determines the level at which the compressor begins to reduce the volume of the signal.

In other words, it's the point at which the compressor becomes active.

To get started, adjust the threshold of the compressor so that the loudest parts of the track are hitting just above the threshold on the compressor's GR (gain reduction meter).

For example, if the loudest parts of the track are hitting around -6 dB on the GR and the threshold is set to -10 dB, the compressor will only be reducing the volume of the signal in the loudest parts of the track.

This will help to even out the dynamic range of the vocal and bring the volume of the vocal up in relation to the rest of the mix.

Step 3: Adjust the Ratio

The ratio of a compressor determines how much it will reduce the volume of the signal above the threshold.

A high ratio will result in a more aggressive reduction of the volume, while a low ratio will result in a softer reduction.

When measuring sound, compression ratios are expressed in decibels.

This means that if a signal is 2dB above the threshold, it will be attenuated down to 1dB.

Similarly, if a signal is 8dB above the threshold, it will be attenuated down to 4dB.

Now, let's look at how certain ratio settings affect a signal.

1:1 — Regardless of the threshold level, the input and output levels remain the same. So, there'll be no compression applied.

1.5:1 — This is a gentle, transparent compression ratio that will preserve the natural sound's peaks and valleys.

2:1 — By using this ratio, the dynamics are smoothly controlled without noticeable changes to the tone or punch of the track.

3:1 — This ratio setting applies moderate compression with a bit more aggressive control. It preserves natural dynamics while applying a good amount of control.

4:1 — Compresses information at a medium ratio, which has better control. There will be a slight difference in punch, loudness, and tone.

10:1—This ratio works aggressively, reducing dynamic range and reducing the punch, clarity, and presence of a signal.

20:1 to Infinity:1—It is limiting because the compressor effectively prevents the signal from crossing the threshold when you start applying this amount of ratio.

Controlling Dynamics: 4:1 to 6:1

Tonal Compression: 1.5:1 to 3:1

Reducing Transients: 5:1 to 8:1

If you're going for punch: 3:1 or less

By the way, save these ratio settings somewhere because they apply to everything, not just vocals.

Step 4: Adjust the Attack and Release Times

The attack and release times of a compressor control how quickly the compressor responds to the volume of the signal.

The attack time determines how quickly the compressor kicks in after the volume of the signal exceeds the threshold, while the release time determines how quickly the compressor lets go of the signal once it falls back below the threshold.

Adjusting the attack and release times of the compressor can have a big impact on the sound of the vocal.

A fast attack time will cause the compressor to kick in quickly, which can help control the dynamics and clarity of the vocal.

A slower attack time will allow more of the initial transient of the vocal to come through before the compressor kicks in, which can help preserve the natural dynamics of the performance.

The release time is equally important, as a fast release time will cause the compressor to let go of the signal quickly, which can help add punch and clarity to the vocal.

A slower release time will allow the compressor to hold onto the signal longer, which can help sustain and smooth out the dynamics of the vocal.

Step 5: Apply Makeup Gain

After you've finished adjusting the threshold, ratio, attack, and release times of the compressor, it's important to apply makeup gain to the compressed signal.

Makeup gain is simply the process of increasing the volume of the compressed signal to bring it back up to its original level.

To apply makeup gain, start by playing back the vocal track and finding the loudest part of the song.

Once you've identified these areas, adjust the output level to match the before-and-after compression.

Keep switching the compressor on and off until the levels are the same.

This will bring the volume of the compressed vocal back up to its original level, ensuring that it sits well in the mix and doesn't sound too quiet or too loud in relation to the other tracks.

Vocal Compression FAQ

Q: How much compression should I use on vocals?

A: There is no one-size-fits-all answer to this question, as the amount of compression you use on vocals will depend on the specific voice and recording.

However, a good starting point is to aim for a ratio of around 3:1 to 4:1, with a threshold that is set just above the loudest parts of the track.

From there, you can make adjustments as needed to achieve the sound you're looking for.

Q: Should I compress the entire vocal track or just certain parts?

A: It's generally a good idea to compress the entire vocal track, as this will help to even out the dynamic range of the performance and bring the vocal up in the mix.

However, you may find that certain parts of the track benefit from more or less compression than others.

In these cases, you can use an automation curve to adjust the compressor's settings on specific sections of the track.

Q: How can I make sure my vocals don't sound too compressed?

A: One way to avoid over-compressing vocals is to use a slower attack time, which will allow more of the initial transient of the vocal to come through before the compressor kicks in.

You can also use a slower release time to help preserve the natural dynamics of the performance.

Additionally, it's important to apply enough makeup gain to bring the volume of the compressed vocal back up to its original level, as this will help to ensure that it doesn't sound too quiet in the mix.

Conclusion

Compressing a vocal track is a crucial step in the mixing process, as it helps to even out the dynamic range of the performance and bring the vocal up in the mix.

It can also be used to add punch, clarity, and sustain to the vocal, as well as shape and control the tone.

By following the steps outlined above, you should be able to effectively compress a vocal track and get it to sound great in your mix.

Remember to use your ears and make adjustments as needed, as every voice and recording is different. With a little practice and experimentation, you'll be a pro at compressing vocals in no time!

Multiband Compression on Vocals

If you're familiar with broadband (single-band) compressors, then using the multiband compressors won't be a problem.

The parameters are mostly the same; the interface may be a bit intimidating at first glance, but it won't take long to get familiar with it.

Multiband compressors are mostly used in mastering, but they can also be used for mixing vocals.

Especially if you have a stack of vocals and only want to compress the high midrange frequencies while leaving the other frequencies alone, or if you just want to fix a problem frequency.

Multiband compressors are good if you're fixing a problem like a nasal frequency, for example, or a percussive part that keeps jumping into the mix.

I wouldn't recommend it as a go-to plugin for vocal processing because it can mess up the timbre of the vocal and change its character.

But if shaping and controlling the timbre is your goal, then go for the multiband, or even if you just want to change the character of the vocal.

A good example would be when the singer keeps looking at the people in the control room while he or she is in the vocal booth.

That usually changes the frequency response of the vocal a bit. To keep the tone consistent, you can use a multiband compressor.

A multiband compressor can be a great tool to make a “dynamically controlled” boost in the high frequencies, for instance.

It’s a good tool for mixing vocals, but use it with caution.

Guide For Using The De-Esser

Vocal de-essing is the most overlooked process in vocal mixing by both novice and experienced engineers.

I've also heard a lot of music coming out of major record labels with plosive and sibilant peaks.

Some even believe the sibilance is a natural thing and there's no need to remove it.

Sibilance and plosive peaks can be annoying to the listener, especially if they're using headphones or earbuds to listen to music.

The aim is not to remove them completely; just bring them down to a more listenable level.

Once you have the vocal compression done, you’ll notice there will be some sibilance on the vocals. Even a good recording will have sibilance, especially if it’s a female vocalist.

These are sounds with “sss” or “ts,” and these are caused by words with alphabets like t, k, s, and z. They’re commonly known as hissing sounds.

These are not generally bad for the mix, but in some cases they can be annoying and can sound pretty obvious after adding effects such as delay.

A de-esser can also degrade vocal clarity, so you should use it but not completely remove the peaks. You need a little bit of sibilance to keep the vocals natural-sounding.

A de-esser is also a dynamic processor, so you’ll need to add it right after the EQ and compressor but before you add any time-based effects like reverb.

This is just to ensure that the de-esser is only working on sibilance and not messing with other frequencies.

How To Get Rid of Sibilance

There are two methods for reducing sibilance in a vocal performance.

You can either use automation by manually gain riding all the sibilance on your vocals, which will take years to finish. But some engineers just prefer doing things manually instead of relying on tools.

Or you can use a dynamic processor called the De-Esser.

A de-esser is a special type of compressor that is designed to remove these sibilant and plosive sounds. A de-esser is simple to use because it typically has three parameters.

Which are sidechain (or listen), range, and a frequency control parameter. The frequency control allows you to specify the frequency at which the de-essing should occur.

Each and every vocalist has a different frequency at which the sibilance occurs.

For instance, a male singer has a low tone, so the sibilance are mostly found around 3-6kHz. Female sibilance are mostly found at 5-8kHz.

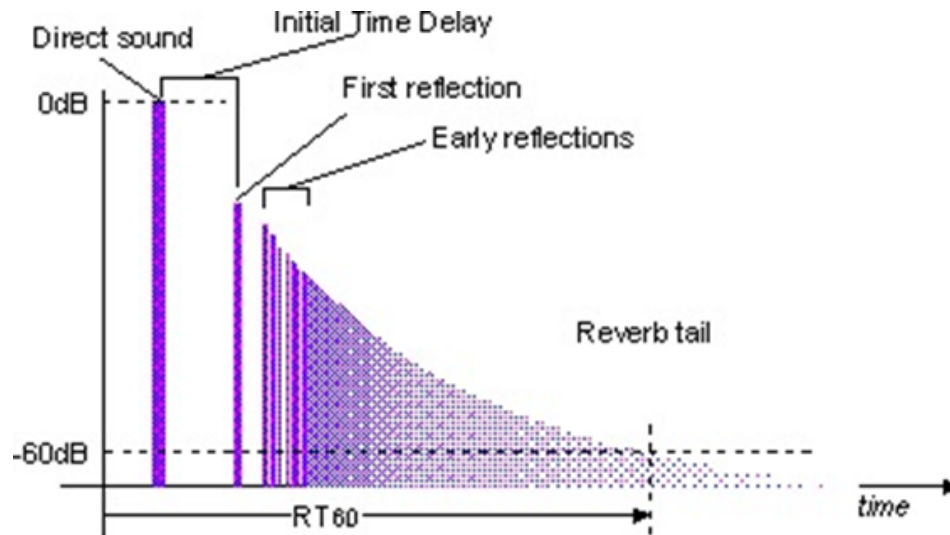
The range is the amount of attenuation applied to whatever frequency you choose, which is mostly at the top end of the frequency spectrum.

The sidechain or listen parameter allows you to listen to the particular high frequencies where the sibilance are happening.

You don't want to use an equalizer for this because an EQ will reduce the sibilance throughout the entire mix, which may sound undesirable or unnatural, and it can make the whole vocal mix sound dull.

So if you're not doing it manually, then use a de-esser dynamic processor.

Reverberating The Vox



A common mistake people make with reverb is to focus on how it sounds instead of how it makes them feel.

Reverb doesn't only add depth or soften the vocals; it also adds emotion.

Choosing a good reverb sound for your vocal is very crucial.

The same way that a multiband compressor can mess up the timbre, the same thing can happen if you choose the wrong reverb sound for the vocalist you're working on.

Try different reverbs until you find one that doesn't change the character of the sound.

There are different kinds of reverbs.

- A room reverb is really short, and it will add little depth and space to the vocal.
- A hall reverb tends to be long, it sounds full and has more reflections.
- A plate reverb carries a lot of early reflections and it has a bright and wide sound. It's also a bit thick but for a short period of time as compared to a hall reverb.
- The chamber is really smooth and thick, it is also low-mid focused so good for someone with a higher tone voice.

For the lead vocal, I mostly go for the plate to keep the vocal upfront and give it a bit of width.

For the supporting vocals (backings, harmonies, etc.), a hall or chamber is usually a good choice to push them behind the lead.

Once you've found a good-sounding reverb, you must determine the pre-delay time that will work well with your song.

Unfortunately, I can't say that you must always use a predelay.

However, in most cases, a reverb will make a sound lose its definition if it's not separated from the direct signal.

So use a predelay to allow the attack of the vocal to kick in first before any effect takes place.

Once you have the pre-delay setup, you can determine the decay time that will fit with the music you're working on.

Reverb effects are really easy to get familiar with, unlike dynamic processors.

But the most important thing is to find the right reverb time settings that will match the tempo of the song. Just make sure that the reverb ends before the next vocal hit or word.

You can do this by using your ear or a simple formula.

- Take 60,000 and divide (\div) it by the tempo of the song.

For instance, if you're mixing a song that's playing at a tempo of 90BPM then divide 60,000 by 90 to get 666.7 milliseconds (0.6 seconds).

One trick I learned in music production school that I don't see a lot of engineers do is automate the reverb throughout the whole arrangement.

For instance, the chorus part will be more reverberated as compared to the verses and other parts.

This can sound really good, especially with delay effects, but don't exaggerate it; it doesn't have to sound obvious to the listener, so use it with caution.

Another great trick, especially if you don't want the vocal to sound like it's in a room, is to remove all the early reflections and only use the tail of the reverb.

That way, the vocal will sound as if it's dry, and it will be upfront in the mix, but the tail will be reverberated.

Don't forget to use the pre-delay time to determine when the tail starts getting reverberated.

This will keep the vocals present even though the attack of the vocal will sound dry and the tail will have reverb.

To help your vocals stay present and never be pushed too much to the back in a mix, use less high-end information in your reverb.

This can be accomplished by creating a simple high-frequency shelf at 10 kHz.

If you use too much reverb, it will push the vocals to the back, and too little will push them upfront in a mix.

Get a good balance that won't ruin the vocal performance—just enough to help the vocal become one with the rest of the mix and keep it organic.

Reverb adds emotional impact to vocals, so don't focus too much on how it sounds but instead on how it makes you feel.

Spend some time tweaking the reverb to get the best sound; never rush.

Delay Effect on Vocals

A plugin delay effect basically records the incoming data, which is the vocal in our case, and stores it in a buffer. whereas older school units relied on tape or digital sampling technology.

Delay effects can be used in a simple form or in more complex patches that involve adding effects such as distortion, an auto-filter, or even an EQ to change the character of the delayed signal or just to clean things up.

Delay effects can be a great way to create a doubling effect on vocals.

Most rock engineers use the delay effect instead of the reverb to help the vocal sit well in a mix without pushing it back or making it sound distant.

Most delay effects will have a way to set the timing of the delay, which is very crucial.

Some will come with a sync tempo button, while in some cases, you'll have to use your ear or simply take 60,000 and divide it by the tempo of the song.

The aim of using a delay effect on vocals is not for it to be heard, but just enough to support the vocal and make it sound bigger.

But never make the delayed signal louder than the original signal.

Short delay times of about 80 ms work well to blend the vocals with the entire mix, especially if the reverb is making your vocals sound too thick and ruining the clarity, then you can use a short delay instead.

Be careful with using long delay feedbacks; they'll add some muddiness.

If you're using the delay effect on an FX channel, then keep it at 100% wet, and if you're using it as an insert, then 30% is a good starting point, then play around with it till you find a good spot.

The most commonly used delay is the "ping pong," which is alternating echoes that are panned hard left and right in the stereo image.

To achieve this effect, you need to make the delay time on the left half of the right side's delay time.

Then the vocal will bounce around the stereo field, from the center (original sound) to the left, then to the right channel.

If a stereo delay effect is adding mud to the vocals, go for a mono delay. Alternatively, you can use a reverb with a long pre-delay time of over 120 ms.

Another great-sounding delay trick is the ducking delay effect. To achieve this effect, you'll have to add a compressor to the delay Fx channel and set the sidechain input to be the vocal aux send.

Use a fast attack and slow release time, then set the threshold and ratio to taste.

The compressor will close the delay when the vocals are playing and open as soon as the vocal compression goes to rest; then the last phrase of that vocal part will echo.

Finally, if you're using long delay times, don't add them throughout the whole song; use them in different parts of the mix or in certain phrases.

Quick Panning Tips For Vocals

This section is going to be the shortest because it depends on the material you're working on. But basically, you want to keep your lead vocals in the center, especially for the verse.

The chorus part needs to be wide, which is why it's recommended to record many takes and pan them left and right. I usually keep everything below 80% pan, I never go to 100%.

Panning one vocal part hard left and the other hard right (100%) is like having a single mono channel except that it will be 3dB louder.

It used to be good to pan things hard, especially for background vocals, to make them sound like they were on the edge of the speaker.

People are using earbuds, so making your vocals too wide might not sound good on these earbuds. Go for something a little tighter, not too wide; anything less than 80% pan will do.

Pan the chorus, stacking vocals according to taste, but make sure none of your vocals disappear when the music is played in mono.

If the stacked vocals were recorded like a choir where you have the tenor, soprano, and bass section, then keep the voices with a

low tone in the center and high harmony vocals on the sides, and if there are Adlibs, keep them in the center.

If all the stacked vocals sound the same, then pan them according to what you feel sounds good. Things you can pan are mostly the harmony vocals and backing vocals.

Experiment with panning and make sure to test your mix in mono.

Mixing Background Vocals

Just like the word says, "backing vocals" means they need to be at the back. Your effects and dynamic processor settings for the backings will differ from the lead vocal settings.

For instance, the compressor will have a fast attack with a medium-to-long release because if the release is short, then the backing vocals will be energetic and loud.

That's not what you want for the backings, and that's why you must use a long release and fast attack to keep them punchy but never loud.

Allow for minimal dynamics in your backing vox; use drastic threshold and ratio settings while avoiding any pumping effect.

You'll also need to use a lot of reverb to push them toward the back of the mix, being careful not to drown them in reverb.

Long reverb and delay times work well because backings don't play throughout the whole song.

Use a stereo image effect to widen up the backing vocals in the mix; this will make them less direct, which will help the lead vocal to lead.

Don't over-process the backing vocals; remember that they need to support the lead vocal, so they shouldn't be pushed too far back, and they also need to be audible and clear.

For the EQ settings, I normally use a drastic low-shelf filter at around 150kHz and a big dip in the low mids.

Sometimes a dip around 1kHz does open up some space for the lead, then I'll add some sparkle using a high shelf. This high-shelf will be a bit more than the lead, though.

Push It To The Limit

A limiter is another great tool you can use for compression.

Unlike a compressor, which acts on a sound as it arrives, a limiter has a look-ahead feature that tends to make the compression a lot smoother even if it's pushed hard.

A limiter usually uses a fast attack, mostly a hard knee, and an unlimited ratio. It's a special type of compressor that will never allow the input signal to exceed the threshold.

Sometimes using a compressor can squash the vocal too quickly and cause it to distort, and in that case, the limiter is a good option as it has a look-ahead feature that allows the limiter to see a few milliseconds before the compression takes place.

Usually, a limiter should be the last thing you add to your vocal chain after all the processing is done. Use it to trim out the loudest peaks or if you want to push your vocals to the front of the mix.

Sometimes you might find yourself using heavy compression with lots of gain reduction, a fast attack, a high ratio, and a lot of make-up gain.

In that case, technically, you might be using the compressor as a limiter.

So try a limiter instead and see how it sounds. You might find that the limiter is working better because you weren't using the compressor for what it was designed to do.

Using a limiter is much faster and easier, especially if you're not familiar with compressors. The vocals will be right in your face without fluctuating.

Use a limiter to help the vocal sit on top if it's getting lost in some parts of the mix.

It's little things like this that make a big difference in the mastering stage, as none of the peaks will keep jumping in the mix.

Which will make it a lot easier for the mastering engineer to polish the mix.

Remember, less is more, so don't squash the voice.

Wrapping Up

Producing great-sounding vocal mixes is all about experimenting. Things like tuning your vocals can also help the vocal blend well with the entire mix.

I would advise you to get the timing and correct vocal pitch from the source instead of relying on tools.

There are a lot of great tools out there for fixing vocal pitch and timing issues, but the most popular ones are Melodyne, Waves Tune, Cubase VariAudio and AutoTune.

Another great tool that can add some sparkle to your vocal mix is the Exciter. However, I don't use an exciter on the lead; I usually use it on the backing vocals or the entire vocal group channel.

Use modulation tools such as chorus to double up sounds.

Effects such as phaser and flanger can add drama to a song when used in places such as the breakdown of a mix.

Modulation plugins work really well on background vocals as well.

SIDECHAIN COMPRESSION ON STEROIDS

Sidechain compression is a really great technique that will help your sounds sit well in a mix without fighting for space in the frequency spectrum.

The technique has gotten a bad reputation over the past few years because of EDM and the dance music scene.

These days, when you mention sidechain, people automatically think kick and bass.

However, sidechain compression, when used right, can take a good mix and make it sound great.

Since this technique is dependent on what plugins or DAW you use, I'll just share the techniques, and your job will be to find out how to apply them with the tools you already have.

Each and every DAW is different, and there are too many plugins that have this feature that it will take ages to explain each of them.

The best thing I can do is share the techniques.

But first, let's check out what sidechain is and what it's all about.

What is Sidechain Compression

The term "sidechain" is mostly associated with the pumping effect.

This was a technique that was only used on radio to duck the music whenever the radio presenter started talking so that the listeners could hear the presenter without the music interfering.

The technique was simply called "ducking" back then. In ducking, the level of one audio signal is reduced by the presence of another signal.

Sidechain made it easier for radio producers and presenters to avoid ducking the music manually each time the radio presenter or DJ speaks.

Then later on, audio engineers started using the technique in pop records and dance music.

Over the past decades, the technique has been used to achieve some great stuff, which I'm going to share with you in this module.

So fasten your safety belt.

Classic Pumping Sidechain Effect

This is the classic and most famous way to use sidechain compression.

The most obvious is to make your bass duck each time the kick drum hits. This is mostly used in EDM, pop, and dance music to avoid the bass clashing with the kick drum.

It also created a pumping effect, which adds movement to the bass sound.

The same technique can be applied to strings and pads, but in this case, it's not used to create space for any sound but to add movement to a boring sustained string section or pad sound.

The pumping effect can also be used in different sections of your arrangements to add some excitement or keep the listener bumping their heads.

To achieve this effect, you can use your original kick drum sound to trigger the other sounds, or you can simply create what's known as "Ghost Notes."

The problem with using a mix's kick drum is that the kick drum may drop in some sections of the song, causing the pumping effect to stop.

If that sounds good and that's what you're going for, then you can use the kick drum in your mix.

However, in some cases, you want the pumping effect to happen throughout the entire mix. In that case, you simply have to use Ghost notes.

Search how to use ghost notes in FL Studio, Cubase, Pro Tools, etc. since each and every DAW is different.

Google is your friend.

You don't need any fancy tools for this; you can use what you already have.

Sidechain Reverb & Delay Technique

The sidechain reverb and delay effect is used to add definition to a mix and make sure that the reverb or delay doesn't mess up with the attack and punch of a sound.

Effects are a good way to polish a mix, plus they'll improve the width and depth, giving it a great three-dimensional feel.

But they can create clutter in a mix or mess up the groove of a song, and that's when sidechain compression can really come in handy.

This technique is mostly used on synth leads and vocals to get a clean and polished sound.

It's really easy to do this; simply add a compressor to your reverb or delay FX channel and adjust the compressor to taste.

The compressor will duck the effect (reverb or delay) whenever the original signal is playing and release the effect when the original signal drops.

It's a cool way to keep your sounds punchy and present. You can do this with any effect that adds clutter or muddiness.

Don't make it too obvious though; keep it subtle since you do need those effects to make your sound sit well in the mix.

You don't need to duck the effect completely, just enough to avoid clutter, or you might lose the 3D feel and end up with a dry and dull mix.

Multiband Sidechain Effect

For most beginners, it can be hard to create space in a mix and avoid sounds clashing for the same space in the frequency spectrum.

In this module, I'll show you how to avoid frequency overlap by using sidechain multiband compression.

Usually, the vocals will clash with the guitars in the upper midrange frequencies, and sidechaining is the best solution for this if EQ and volume balance have failed.

Don't worry; you don't need any fancy multiband compressor to do this. Every reputable DAW can achieve this technique.

To help your vocals shine in a mix without the guitar interfering, you simply use a multiband compressor in the upper. This clash usually happens around 1kHz to 2.5kHz.

The trick is to make the guitar duck a bit in that frequency range whenever the vocal is playing in the mix.

This doesn't have to be guitar and vocal only; it could be any sounds that are fighting for the same space in the frequency spectrum.

Only use this technique if EQ and volume balance have failed you.

Remember, the listener shouldn't hear the pumping. If they can hear the effect, then you're doing it wrong.

You just need enough to avoid frequency overlap and help a particular sound shine in a mix.

Gated Sidechain Effect

Gating isn't just useful for rhythmic chops and stutters or to remove unwanted noise.

You can create interesting effects by placing a gate after a reverb, delay, or chorus effect on an FX channel and then triggering the gate to open only when the signal being sent to the bus is playing.

This can make for powerful effects on vocals, synths, guitars, and many other types of signals.

You can even leave a little tail on your reverb by extending the release and/or hold times.

Automate The Sidechain

Another great technique you can use to improve the arrangement of a mix is to automate the pumping effect.

This will give you some seriously interesting arrangements and keep your listener glued to the stereo.

You can automate anything here. Maybe automate the threshold to add more pumping or less pumping in some parts of the song.

Or even play around with the release time to create a lazier feel and add contrast to certain parts of your arrangement.

The possibilities are just endless and depend on taste.

Wrapping Up Sidechain

This technique can be used for many reasons.

You can even sidechain a distortion or saturation effect to create interesting rhythmic drive or dynamically distorted tones.

I hope this module helped you realize how powerful sidechain can be, and you'll realize that it can be used for many reasons besides ducking the bass from the kick.

Simply be creative with it, and don't be afraid to experiment.

MIXING ACOUSTIC GUITAR IN 3 EASY STEPS

In this tutorial, I'll show you what to do after recording an acoustic guitar to help it sit well in the mix and cut through.

The first thing you need to look at before mixing an acoustic guitar is whether it's a rhythm section or just a solo.

If it's a solo, then use a reverb and delay to push it toward the back of the mix.

If it's the rhythm section, then you can choose to push it a bit upfront, but make sure it doesn't clash with or overpower the vocals and other sounds in the mix.

The problem a lot of people face when mixing an acoustic guitar is getting the right level, frequency balance, EQ, and compression.

So, to assist you in resolving this issue, I will share three tips for mixing acoustic guitars.

Recording Acoustic Guitars

If it's not good from the source, then you're going to have a problem when mixing.

If your room is not acoustically treated, then I would advise you to buy an acoustic guitar pickup instead.

If you have both a microphone and a guitar pickup, try both and see which one gives you good results.

Always record in mono for a punchy and clear guitar sound; this will also help avoid phasing problems.

Acoustic guitars, like brass instruments, always sound great when performed live.

Some good plugins like Native Instruments are trying hard to bridge the gap, but I still prefer recorded guitars as compared to MIDI-programmed ones.

Keep It Natural

Find a good balance for the acoustic guitar in the mix by using the level fader.

If you still struggle to get it to sit in the mix by adjusting the volume, then that's when you'll need processing.

If you have a busy mix, then you might need some heavy processing, but if the music doesn't have a lot of sounds, then keep it as natural as possible.

Processing or Mixing Acoustic Guitars

If you can't find a good balance for the guitar, then you'll need some processing to help it sit well in the mix, especially if it's a busy mix.

If your song has instruments dominating the low-end, such as the kick and bass, then cutting everything below 100Hz on the guitar might make it sound thin.

Use a low-shelf only if the acoustic guitar has some low-end rumble, sounds boomy, or adds low-end mud.

If the acoustic guitar is boxy or boomy, then a cut around 100Hz – 300Hz will get rid of the boominess.

Cutting from 1kHz-3kHz will make space for the vocals and make the sound more transparent and open.

To help the guitar cut through the mix, a boost at 8kHz-10kHz will do the trick. You can also boost the 5kHz-7kHz range to add presence to the guitar.

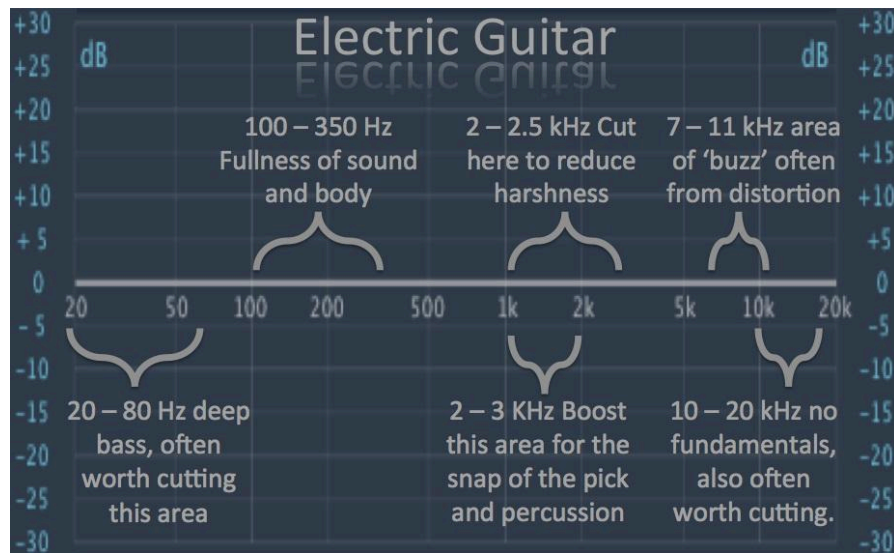
Use effects such as reverb as a tool to push the guitar upfront or back in the mix. Also use a distortion effect or bitcrusher to add some dirt and warmth to the guitar.

I would avoid compression by any means necessary when it comes to mixing acoustic guitar. Try to stick with distortion, overdrive, or saturation.

Do some manual gain instead of using compression.

The key to mixing acoustic guitars is to **keep them sounding as natural as possible.**

TIPS FOR MIXING ELECTRIC GUITAR



I'm happy to finally write about one of my favorite instruments.

This sound is in most of my productions; yes, I'm talking about the electric guitar, and below you'll learn how to mix electric guitars using EQ, effects, and dynamic processing.

One thing I've realized is that mixing an electric guitar differs from song to song; there's no set pattern.

Each time the effects will need different settings—different amounts of compression, width, distortion, and EQ.

Now let's look at how you can mix an electric guitar the traditional way without the CLA or any other Waves Signature plugins and still get a good sound.

The first thing we'll look at is equalizing.

Mixing Electric Guitar Eq

No matter how the guitar is recorded, all instruments have some elementary and fundamental areas you need to listen out for.

I won't get into recording; this module is all about mixing the guitar.

The electric guitar has a lot of energy in the mid-range frequencies. An electric guitar can also carry a lot of bass frequencies.

More especially if the guitar is playing chords, then it might clash with the bass guitar, but not only in the low frequencies but also in the lower mids.

Which in result will *make your mix muddy*.

A dip at around 100Hz to 150Hz will fix that masking problem.

If your guitar is playing percussively (perhaps as a solo), it will almost never create masking in the lower mids.

But that all depends on what you want to achieve (your end goal).

If you're looking for a more deep-bass electric guitar, then you'll need to boost around the 100Hz range.

If your guitar needs some warmth, a boost at 250Hz will either add warmth or add muddiness. So you need to be careful and know what you want the guitar to sound like.

Here's an EQ chart to help you know what you're adding or removing when equalizing an electric guitar.

Bottom end: 100Hz

Warmth: 250Hz

Mud: 200Hz – 350Hz

Body: 500Hz

Pick Sound: 1kHz – 2kHz

Harshness: 1kHz – 3kHz

To Push Guitar Upfront Boost Around: 3kHz – 4kHz

Presence: 5kHz

Clarity: 10kHz

This might not be 100% accurate, but at least it will give you an idea of where to start.

If you think your guitar is too harsh, then by looking at the EQ chart above, you'll know where to cut. If it's lacking presence you know where to boost etc.

Now let's look at dynamic processing.

Compression Settings For Electric Guitar

If your guitar is heavily distorted, then it already has some compression on it, and even if you check the waveform, you'll see that it's a brickwall.

Then, in that case, you'll just add minor compression to help it sit well in the mix.

To avoid overcompressing the guitar, try some parallel compression as well.

But if the guitar has more dynamics and the volume is constantly changing, then you'll need more compression.

If it's percussive, then use a fast attack with a medium release time. If you just want to tame out the loud peaks and keep the sound punchy, then use a fast release time.

The ratio and threshold settings will depend on how large the dynamic range is.

I've covered compression settings in previous modules, so we'll skip it for now.

And if you're mixing a stack of guitars, then use a multiband compressor instead of a single-band compressor. You'll be able to control the dynamics a lot easier.

Modulation and Effects

Distortion, overdrive, or saturation will add a lot of weight and warmth to an electric guitar mix.

But note that heavily distorted guitars can also clash with the vocals; they will make your vocal mix sound muddy.

If, in your case, you need to remove distortion instead of adding it, then use an EQ to cut out frequencies at around 7kHz.

Distortion can sometimes be called "overdrive."

But overdrive is different; overdrive has a mild effect with a lower gain that is suitable for blues, jazz, and classic.

Distortion produces harder and harsher sounds, which are suitable for hard rock and metal.

A distortion plugin can also change the timbre like an EQ because it's also some form of amplification. So be careful when you distort your electric guitars, and remember that **less is more**.

Flanger and Phaser are really good ear candy for an electric guitar.

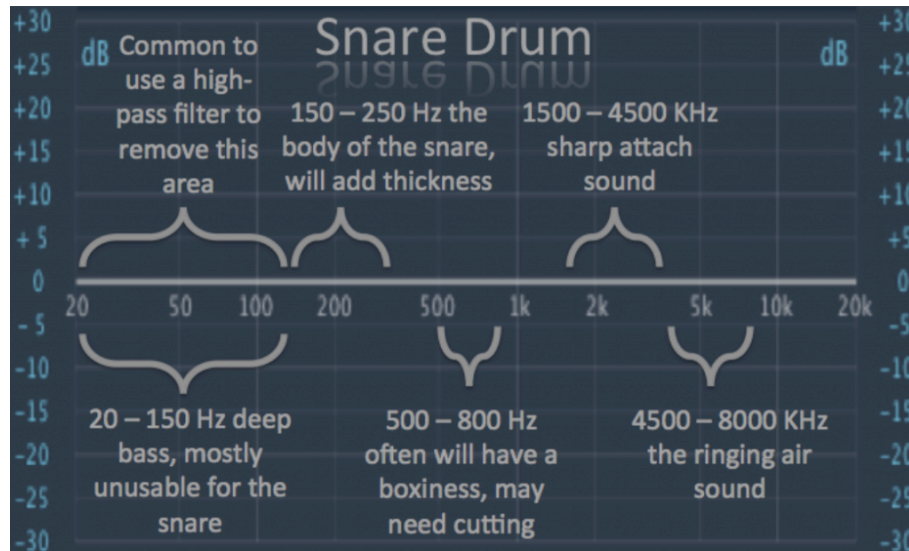
But they sound dope when introduced in some parts of your song to add some drama. I don't add these throughout the whole song.

The Chorus effect is also a good tool to double up your guitar, but it can also push it to the back of the mix when you add too much.

To get a good stereo balance, use reverb, stereo image tools, and delay.

That will help the electric guitar find its space in the mix.

SNARE EQ



In this module, I will be sharing with you a snare EQ guide to help you get a phat and punchy snare using an equalizer.

But before you start adding EQ to your drums, check if there's any phase cancellation and tune your drums.

Check Phase Relationship

The first thing you need to look at is whether the snare was recorded live or if you are using samples.

If it's a live snare, the first thing to do is check its phase. If there's phase cancellation, then the snare won't be punchy.

People believe that phase only occurs when using live recorded drums, but this is not the case; it can occur with layered samples.

This can also result in phase cancellation, so make sure you check the phase relationship of the snare drums to see if they work well together and benefit the entire mix.

Another thing to check when layering is the key, length, and starting point of each snare.

Make sure that all the snares you chose are in the same key, have the same length, and all hit at the same time. Each snare should be assigned to its right frequency (low, mid, and high).

For live snare drums, check out the phase tutorial above.

Snare EQ Guide

As you know, there are a lot of different snare drum sounds out there.

So there's no one EQ setting that will work with all snare sounds.

You'll have to be hands-on to see which EQ settings will work best for the material you're currently working on.

But each sound has certain fundamentals, and this EQ guide will help you understand where to cut or boost to help your snare drums sit well with other sounds in a mix.

If your snare drum is too wimpy or weak, then give it a small boost around **60Hz-120Hz**.

Below **60Hz** is where you'll find the rumble on acoustic snare drums, so make sure you cut that out using a high-pass filter.

Usually all samples come without any rumble, and cutting those frequencies will just result in a weak and thin snare sound.

The warmth of a snare drum is mostly found around **120Hz-200Hz** this part of the spectrum fills out the snare drum. To emphasize the phatness and punch of the snare drum, a boost around **195Hz-250Hz** will do the trick.

To get a snappy snare sound, boost around **6kHz-8kHz** this will also add presence to the snare drum.

250Hz-400Hz is the muddy area of the snare drum.

2kHz to 3.5kHz is where you'll find the crunch of the snare drum.

Also, add some air to your snare to make it shine by making a boost at **10kHz**.

Sometimes the EQ settings are not the issue, but the level adjustment of other sounds is overpowering the snare, so double-check the levels of other sounds.

But if the problem is the snare EQ settings, then this guide should be able to help you fix that.

ADVICE FOR MIXING PIANO SOUNDS

Before getting into this module, I assume you already recorded a good piano sound or chose a good one if you're using VST instruments or samples.

This tutorial is about how to mix a piano using an equalizer and compressor, and how to get good width and depth to make a piano sit well with other sounds in a mix.

Before diving into piano mixing, you need to determine what the role of the piano is in the entire song.

Does it have a percussive touch, or is it playing sustained chords? Are there any other instruments playing, or is it an acoustic song?

If it's an acoustic song, then the piano needs to dominate and become bright.

For instance, a piano sound should be mixed brightly on a POP song, while a dark piano sound works well on a blues or jazz song.

When you know what the end results should sound like, then you'll have won half the battle.

If you know what the piano should sound like, you have a good advantage. If you don't, then listen to other related songs to get an idea.

Piano EQ Settings

As always, the best practice for getting good EQ settings is to use a bandpass filter and sweep around the frequency spectrum to find all the problem frequencies and cut them out or boost them where necessary.

Below, I have a piano EQ guide or chart that will help you determine the right EQ settings for your piano sounds.

This EQ guide will help you determine whether you are benefiting or ruining the sound when you boost or cut.

Here are the piano EQ settings:

50Hz-100Hz ~ Adds bottom

100Hz-250Hz ~ Adds roundness

250Hz-1kHz ~ Muddiness area

1kHz-6kHz ~ Adds presence

6kHz-8Khz ~ Adds clarity

8kHz-12kHz ~ Adds hiss

If your piano sound is muddy, then cut around **300Hz** that should get rid of the mud.

If it's too thin, a boost around **100Hz** to **250Hz** will add some roundness to the piano.

If you need the piano to be bright, then add some air using a high shelf EQ to boost around **15kHz** to **20kHz**.

Boost till you can hear the sustain pedal sound, then drop it down a bit.

If you're going for something mellow, lower the frequency range to **15kHz** to **20kHz** with a high-shelf filter.

Don't forget to remove all the rumble below 50Hz to avoid any low-end mud if you're working with a live recording.

Always make your cuts subtle when it comes to a piano. The key is to keep it sounding as natural as possible without messing up its pitch and timbre.

Compressing Piano Sounds

When you choose a compressor for a piano, you have to choose one that won't mess up the timbre of the sound. The piano needs to sound as natural as possible.

If the piano is playing sustained chords, then a compressor with a fast attack and a medium-to-long release will mostly work well.

If it's percussive, like a solo, you'll need to use a quick attack with a quick to medium release.

Depending on the dynamic range, a ratio of **4:1** or less should work.

Just add enough compression to tame out all the loud peaks without messing up the tone or timbre of the sound.

Don't make the piano distort by using an attack that is too fast; let the transients go in first, then let the compression kick in gradually and fade out smoothly without creating a pumping effect.

Mixing Piano – Width and Depth

The final step would be to make the piano sound good in stereo. Use panning and other stereo image tools to deal with the sides.

When I'm mixing, I usually pan a piano sound left or right (never in the center) if it's a percussive sound like a solo.

If it's chords, then I prefer keeping it in the center and using a stereo image tool to make it wide instead.

Use the reverb effect to push the piano back or forward in a mix, and use other effects such as delay to create width and depth.

Reverb and delay will help you find a good balance for the piano to sit well with other sounds in a mix. Too much reverb on a piano sound will make your entire mix muddy, so add enough without making the piano washy.

To add more movement and feeling to your piano sound mix, use effects such as phaser, flanger, and auto-filter.

If you decide to add more movement, then use a small amount of reverb to make the piano intimate and add some delay.

TIPS FOR MIXING STRING INSTRUMENTS

Strings are really good for adding that final touch to your productions, but they tend to stick out in a mix more than you need them to.

That mostly happens because strings can have a lot of high-frequency content.

In this tutorial, I will show you how to fix that problem by simply using an equalizer, reverb, and modulation plugins.

High-string sounds such as cello, viola, and violin are not that hard to mix because the ear is sensitive to high-frequency content.

Lower-frequency strings are the ones that tend to be challenging and can make your precious mix muddy, especially if the strings are playing chords.

Now, let's look at what you can do to make your strings sound good and work well with the other sounds in your mix.

Dynamic Processing

The first thing I always do is add compression to tame out the loud peaks and keep the strings at a constant level.

If the music notes are sustained, then I use a fast attack and medium-to-long release.

If the strings are playing short notes, like pizzicato strings, then a fast attack and fast release will work well.

The ratio settings will depend on how loud the peaks are. The louder the peaks, the higher the dynamic range, which means you'll need more ratio to smooth out those loud peaks.

And the lower the peaks, the better a lower compression ratio will work.

Keeping the volume of the strings constant makes the equalization process a lot easier.

However, you don't want to kill all the dynamics; over-compression will just ruin the life out of the strings and make them sound robotic or unnatural.

The more compression you add, the less realistic the sound will be, especially for VST instruments.

All you need to do is make sure there isn't too much difference between the loud and soft parts of the sound.

Equalizer Settings

Keep in mind that low strings won't be mixed the same as high strings.

Let us first look at how to EQ high strings.

If you're mixing high strings, then you will mostly do some boosting in the lower range of the spectrum and a lot of cutting at the top end.

A small boost with a narrow Q at around **100Hz to 150Hz** will add some bottom to the instrument.

A cut around **205Hz** will remove any low-mid muddiness.

To add some body or fullness to high strings, make a boost around **380Hz**, use a wider Q factor. Most of the loud peaks you'll need to cut are around **2.5kHz, 4.6kHz** and **7kHz**.

Cut each of those frequencies using a narrow Q for all. Sometimes you might find that there are more peaks; cut them out if necessary.

For low-frequency, rich strings, the opposite of what I said above will work. You'll start by removing anything below **60Hz**. If the strings are clashing with the bass guitar in the lower mids, a cut around **100Hz to 250Hz** will fix the masking.

Another cut from **250Hz** to around **1kHz** will remove any muddiness.

To help the lower strings sit well in the mix, you'll need to make a boost with a wide Q at around **1kHz** to around **6kHz**. That will add some crunch, but sweep around that frequency range to find the sweet spot.

To add some presence or clarity, a boost around **6kHz** to **8kHz** will do the trick. Finally, making another boost around **8kHz** to **12kHz** will add some air to your strings.

Use these EQ settings as a guideline when you're mixing your strings; they work for me all the time, and I hope you find them useful.

Reverb

Strings always sound better with reverb, whether they are real strings, samples, or from a VST instrument.

In this part of the tutorial, I'm going to help you find the right settings for your reverb to make the strings sing.

Strings sound good when you place them in a medium- to large-sized artificial hall.

If your reverb effect has a hall or church preset, that's a good place to start, then tweak it till it fits with what you're working on.

A reverb decay of four seconds or more works well on string sounds.

A short decay time will usually make things sound bigger, but that's not the sound you're going for; when mixing strings, you want to get a lush sound.

To avoid the reverb from making your strings muddy, use the pre-delay parameter to let the attack of the strings kick in first.

Don't let the reverb kick in at the same time as the strings, and use a long or medium reverb time.

Note that these settings will work well for a string section that is playing sustained or long notes. The opposite of these settings might work well on strings that play shorter notes. Just play around with it.

But without the reverb, your strings will sound dull. Reverb just makes them sound more realistic.

You can also use modulation effects to make your strings sound a lot better. For instance, adding a chorus effect can make your strings sound larger.

If your string section is sounding thin, adding a chorus effect will thicken them up.

But too much can push the strings at the back or make the mix muddy. So use a subtle amount of chorus.

Flanger and Phaser can also be used to make strings sound larger; just be careful not to create a wobbly sound effect if you use a Flanger.

Add one modulation to your string mix; it might be the missing link in your mix. That's how easy it is to get a good string section mix; don't complicate it.

How To Enhance Sounds By Adding Harmonic Excitement

In this tutorial, we'll be looking at a different way of adding harmonic excitement to enhance sounds in a mix. I'm going to show you the techniques I use to glue sounds together and fatten them by using parallel tape saturation.

It is the same as parallel compression; the only difference is that you'll be using a tape saturation effect instead of a compressor.

You can use any saturation effect to add harmonics to your sounds. It can be distortion, amp-simulation, tube, exciter, or overdrive.

For this example, I'll be using the tape saturation plugin on a drum bus and bass.

I'll also show you how to use a harmonic plugin on an electric piano.

This is to show you how powerful saturation effects are and how to use them in your mixes to add harmonics. Saturation effects are really versatile and are not only designed for guitars.

Let's take a look at how to enhance a drum bus by using parallel distortion. You can add compression to control the dynamics of the saturation signal or EQ to shape the tone.

But for this example, I didn't add any other effects; it's just the Kramer Tape plugin.

Check out the drums below with and without saturation.

I did exaggerate the effect a bit so that you can hear the difference, but you only need a subtle amount of saturation to enhance the harmonics and avoid phase cancellation.

Drums Before Saturation:

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-unprocessed.mp3>

Drums After Saturation:

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-processed.mp3>

If you can't hear the difference, it's because the processing is done in parallel, so the dry signal is still dominant, and we're only adding harmonics, not changing the tone or timbre.

Now, let's look at how to enhance a bass sound using parallel tape saturation.

I'll use the same plugin but with different settings; no extra processing will be applied to the saturation signal.

Listen to the before and after below.

Drums & Bass (Unprocessed):

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-Bass-Unprocessed.mp3>

Drums & Bass (Processed):

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-Bass-Processed.mp3>

If you listen carefully, you'll realize that the bass volume comes up without messing up the low-end frequency range.

The tape saturation is increasing or enhancing the harmonics of the bass sound.

That's how you enhance sounds using tape saturation.

As promised, below I'll show you how to use harmonic plugins to enhance an electric piano.

Adding Harmonics To An Electric Piano

In this part of the tutorial, I'm going to show you how to get an electric piano sound to sit well in a mix without using EQ, compression, or any other effects.

For this example, I'm going to use the same drum kit, and there'll be no parallel processing on this one. I'm going to be using the Maserati Harmonics plugin as an insert to enhance the E-Piano.

All I did was increase the sensitivity, push the size to full, and add spread to make the sound wider because it was too monophonic.

Listen to the difference below.

Drums & Electric Piano (Unprocessed):

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-Piano-Unprocessed.mp3>

Drums & Electric Piano (Processed):

<http://talkinmusic.com/wp-content/uploads/2014/09/Parallel-Distortion-Blog-Post-Drums-Piano-Processed.mp3>

As you can hear, this time there's a major difference as compared to the first examples because there's no parallel processing this time.

The plugin is directly inserted into the signal and processes it at 100%.

I trust that by now you can see the power of using harmonic tools to enhance sounds in a mix.

Use saturation, but use it with caution, and remember that less is more. Done the right way, it can add depth, color, warmth, crunch, or additional detail to a sound, as in our first 2 examples.

8 TIPS FOR MIXING MUSIC WITH HEADPHONES

Is it possible to mix with headphones?

There are a lot of yes and no responses with good reasons everywhere you see this question pop up.

Basically, this means it's possible to do audio mixing with headphones, but it will depend on your experience, tools, knowledge, and a whole lot of other factors.

Some musicians don't have the luxury of building a professional studio where they live because they might not have enough space.

Sometimes they have the equipment but can't play it loud because there are kids sleeping.

No matter the reason, some people don't have any other choice but to rely on headphones for both production and audio engineering.

So in this tutorial, I'll share with you some tips that will help you get great mixes using headphones.

Get Good Quality Headphones

It goes without saying that the headphones you'll be using to mix should not be ordinary. And don't use Beats by Dr. Dre.

Those headphones sound good for listening purposes, but they were not designed for audio engineering.

Beats by Dr. Dre have an exaggerated boost around 100Hz which can easily fool you into thinking they sound good.

Instead, you need headphones with a flat frequency response, like the Audio Technica ATH-M50.

These are designed for mixing music; the Audio Technica ATH-M50 are pro studio monitor headphones.

Other companies that make good headphones for audio engineers are Sennheiser, AKG, Shure, Sony and Yamaha.

But the ATH-M50 always comes out on top. So get yourself a good pair of studio monitor headphones that are designed for the purpose of mixing music.

Protect Your Ears

Mixing can be a time-consuming process, so it's a good idea to take regular breaks to avoid ear fatigue or hearing damage.

Keep the overall volume of the mix at soft levels. That will not only help you get a punchy sound, but you'll be able to protect your ears and mix for longer periods.

Ears are always fresh in the morning, and that's a good time to do your mixing. Don't take 5-minute breaks when mixing; that will not be enough time to refresh your ears.

Take a long walk or drive without listening to loud music; that will help. Just take good care of your ears, and they'll take care of you.

| OSHA's Permissible Noise Exposure | |
|-----------------------------------|------------|
| 90 dB | 8.0 hours |
| 92 dB | 6.0 hours |
| 95 dB | 4.0 hours |
| 97 dB | 3.0 hours |
| 100 dB | 2.0 hours |
| 102 dB | 1.5 hours |
| 105 dB | 1.0 hours |
| 110 dB | 30 minutes |
| 115 dB | 15 minutes |

Headphone Experience

Those who use headphones for production, engineering, listening to favorite songs, and watching movies have a better chance of making a good mix.

They know what a good quality song should sound like on their headphones.

Spend a lot of time with your pair of headphones; it is crucial for you to know them in and out.

Getting the Right Balance

Getting all sounds well balanced in a mix using headphones can be a challenge. If you can get the correct balance from the source, then that will make your mixing job a lot easier.

Another thing that makes the whole balancing thing a problem is that there is not much low-end information on headphones; it becomes guesswork.

But if you're used to listening to other people's music with those headphones, then it won't be a big deal to get the mix well balanced.

... and this leads to our next tip.

Use Reference Tracks

If you're using headphones, then you're going to need some material to use as a reference. Particularly in the low and low-mid frequency ranges.

Use reference tracks to make sure nothing is exaggerated or too quiet in the mix, and you have everything well balanced.

Get as many references as you can and make sure they have similar instrumentation, even if it's not the same genre.

Avoiding Stereo Image Problems

One thing I've realized is that if I pan sounds right and left when mixing on headphones, when I test the mix on speakers, most of the sounds that were panned disappear.

This makes the entire mix sound unbalanced, and that's when reference tracks come in handy.

Headphones can make sounds seem louder, so be careful.

Another thing to avoid is heavily relying on stereo image plugins; instead, keep all processing as subtle as possible.

Bad stereo image shaping can make your music lose punch and definition.

Subtle Amount of Processing

When you're using headphones, make sure that whatever processing you add is subtle.

If you use too much delay and reverb, your mix will become muddy and the sounds will lack clarity. So avoid drastic processing effects to keep the sounds punchy.

Too much EQ will change the tone or timbre of a sound.

If you add a lot of compression, your mix will have an undesirable pumping effect, ruin the transients, lack dynamics, and kill the life out of your mix.

Just make sure that you add a subtle amount of processing and keep things dry (not 100% dry though) because the ear cups are too close to your ears, which can lead you to add more processing.

But remember that if the processing is too subtle, some sounds won't sit well in the mix and will lack width and depth, so get a good balance.

Acoustic Simulation

Ask yourself why some people don't use headphones at all, not even earbuds. There are people who never listen to music with any kind of earpiece.

The music may be enjoyable for a short period of time before becoming strange or causing a headache.

This is caused by the **shadowing effect**.

When music is playing through speakers, the sounds from the left speaker are received on the left ear and then travel to the right ear.

The same sounds coming from the right speaker will also be received by the left ear.

But both ears don't receive sounds from one speaker at the same time; on the other ear, they arrive a bit later and at a slightly reduced level.

With headphones, your left ear only receives sounds that are on the left channel, which is unnatural.

That is the shadowing effect.

You can use tools such as acoustic simulation plugins to fix this problem. They feed a bit of the left channel to the right channel to mimic the natural feel.

These may be useful, but they are only tools and are never accurate.

Headphone Mix Translation

This is the final and most crucial part of mixing music: getting your song to translate well on most sound devices.

Once you're happy with your mix, burn it on a CD and listen to it on different sound systems.

To get a good hearing, do some DIY mastering on your final mix, then test it on different sound systems, both indoors and outdoors, alongside your reference tracks.

In most cases, you'll find that you need to make some more adjustments.

That is why I wouldn't advise you to do a final mix on headphones; instead, take it to the studio and do the final touches using speakers.

Headphones are good for exposing things such as pops, clicks, distortion, or over-processing in a mix.

As a result, use them as a reference monitoring sound device rather than your primary audio mixing output device.

Mix with speakers first and test with headphones later, not the other way around.

But as you know, there's no right or wrong way when it comes to mixing and creativity as a whole; these are only guidelines. Just make sure what you do matches industry standards.

TIPS FOR MIXING TOM DRUMS

Toms are a really great way to add texture to drum fills, create section transitions, or even drive the entire song.

In this tutorial, we'll be looking at some tips that will help you get tom drums to cut through a mix.

Make sure that your toms are tuned and recorded well in order to get a good mix. If you're using VST or audio samples, then make sure you choose the best tom samples.

If your tom drums are recorded live or you're layering different tom sounds to create one big tom sound, then make sure there's no phase cancellation.

If they're live, also make sure that there's no mic bleed, or if you want the bleed to be there, blend it well with other sounds and avoid phase issues.

EQ Settings For Tom Drums

Toms are the easiest of all drum sounds to mix. You just need to focus on the **thump, attack, stick, and air**.

That's basically what you need to focus on. The thump is the resonance or warmth of the tom, also known as the "punch."

Higher rack toms don't rely on the thump as much as floor toms and resonant rack toms do.

If your tom drums are lacking thump then make a boost around **100Hz-250Hz**.

The attack brings out the rhythmic nature of the tom and makes it cut through a mix, making it more audible and present.

To add more attack, boost around **3kHz-5kHz**.

The stick is mostly known as the "click of the tom," which you can enhance by boosting around **6kHz-8kHz**.

The air on the toms makes the drums shine in a mix.

Some people may use the overheads to get some air for the toms, but there could be too much mic bleed or spill, so I would advise you to add the air directly to the toms by creating a boost around **7kHz-12kHz**.

The way I equalize toms is by cutting out anything below **60Hz** to remove the rumble (only on a live recording, not on samples).

Then bring up the thump, stick, and attack and add some air.

Finally, make a big dip in the low-mids to remove mud and ringing tom noise.

However, don't do things by default; always boost or cut when it's necessary.

Compressing Tom Drums

Toms don't really need a lot of compression in order for them to sit well in a mix.

But that will all depend on the dynamic range of the toms.

A slow attack of about **10 ms** and a medium-to-long release time will work well.

A ratio of **4:1** or less is good, resulting in a gain reduction of about **-4dB**.

You can also add your toms to a group or bus channel to glue them together and add more processing, then send your tom group to the drum bus after processing them.

Panning and Effects

When it comes to panning tom drums, I refer to how the drum kit was setup and pan according to how the drummer was positioned.

This is called the drummer's perspective; others prefer the audience's perspective. This is more of an artistic choice.

I don't keep any toms in the center of the stereo image; I always pan them right and left.

If you have an overhead track, then listen to the panning and follow that. If you do the opposite, you'll have phasing issues.

When there's no overhead track, pull up a picture of a drum kit and pan using that as a guide.

A reverb effect will help you find space and depth for the toms so that they sit well in a mix.

Use a short room or plate reverb, and don't add too much reverb; it will muddy your mix and push the toms to the back of the mix.

To help avoid the mud, I always add an EQ to the reverb signal.

On the reverb signal, use a high-pass filter to cut out everything below **60Hz** and make a dip in the lower mids that should prevent the reverb from adding mud to your tom mix.

Finally, the last thing I would do is add tape saturation or distortion.

This will add some crunch, analog warmth, and harmonics to the toms. It will also round off the top end to make the toms fit well with the rest of the instrumentation.

Anything else after that is just cherry on top.

LEARN HOW TO MIX LEAD SYNTH SOUNDS

This is a quick tutorial about mixing synthesizer sounds.

Lately, I've been spending some time helping people on forums. I see a lot of people struggling to mix synth sounds.

A lot of people jump for tools such as EQ, compressors, etc. to mix a synth, which might not work because it's the wrong approach.

When I saw someone asking for a synth EQ chart, that's when I knew I had to create a tutorial about this issue.

When you use a synth, you must build a good sound from the ground up. You have to create the type of sound you want from the source.

This is where reading the manual really comes in handy.

The key here is to know your synthesizer inside and out.

You have to know how to create a good-sounding synth by tuning the synth's envelopes, LFOs, resonance, oscillators, cut-off, and filters without relying on any third-party processing tools.

This will take more than just tweaking a few knobs and then hoping for the best.

It is more about making the sound you want rather than looking for it.

You need to spend time learning how to use your favorite synthesizer.

Synth EQ Guide

I would advise you to avoid using third-party plugins to process synth sounds.

Always strive to get it right from the source by using the sound design and processing tools that come with the synth.

Yet we have to realize that not all synths are built with good processing tools.

In that case, you will need to use a third-party equalizer. Below is a synth EQ guide I created myself by listening to different types of synth sounds. I hope it helps you.

Warmth/Bottom: 100Hz – 250Hz

Muddiness 250Hz – 800Hz

Presence 1kHz – 6kHz

Clarity 5kHz -7kHz

Sharpness 7kHz

Top end 10kHz

If your sound is sounding too digital or fake, give it a boost at 15kHz. This boost usually adds what's missing from digital synthesizers to make them sound more realistic.

Compression Settings

A third-party compressor will suffocate your synth sounds, especially analogue synthesizers.

To deal with loud peaks, **use the envelope** or built-in compressor rather than third-party compressors.

My teacher, back in music production school, used to tell us that using EQ and compression on a synthesizer is a lazy man's job.

Even the sound you'll achieve won't be as satisfying as using envelopes and filters.

So What Do I Do?

Spend some time reading the manual of only one synth that you use a lot, **learn it, and master it**. Once you've mastered one, then the other synthesizers will be a walk in the park.

You can use third-party processing tools; no one is stopping you, but just know that you could get better results with proper attention to sound design.

Third-party effects I would recommend you use are reverb, delay, distortion, or any tape saturation tool. Especially an outboard distortion effect.

Everyone who'll tell you otherwise either has little knowledge about sound design or hasn't tried this approach.

Or even worse, **they tried it for 3 minutes and gave up.**

I'll be brutally honest with you: depending on your experience, it can take a while to design a good-sounding synth from scratch or build it from a preset.

So be prepared to spend time on the sound design part and forget about fixing stuff in the mixing stage.

I hope you understand what I'm telling you here **because this is very important**, and this also applies to synth bass sounds.

But at the end of the day, it's all about taste or preference, so do what works best for you.

How To Mix SAXOPHONE

Here's a step-by-step tutorial that will teach you how to mix a saxophone. I'll be using an alto sax for this tutorial.

Brass and woodwind sounds always sound better when they are recorded live as compared to using a VST plugin.

On this module, I won't talk about recording or programming; just make sure the saxophone is well recorded and already sounds good from the source.

If you're ready and believe you've captured a great sound, then we can move on to mixing.

Check out the audio we'll be working on below:

Unmixed Sax:

<http://talkinmusic.com/wp-content/uploads/2014/10/Mixing-Sax-Blog-Post-Unmixed.mp3>

As you can hear in the audio example above, the alto sax was well recorded and doesn't need much mixing.

That's exactly what you have to strive for if you want your mixes to sound professional. Get it right from the source.

The first thing I did was add compression and an EQ, followed by some effects to blend it with the other sounds in the mix.

Saxophone Compression Settings

When you have a well-recorded signal, you won't need much compression, which helps the sax sound more natural.

For brass sounds, I prefer to use mild compression just to keep the volume constant throughout the whole mix.

The saxophone I'm working on was recorded in mono; I don't like recording in stereo because it can cause phase cancellation issues, and I want my sounds to be clear and punchy, not just bright.

I added a **mono compressor** by FabFilter, but the compressor is not doing much for the sound.

It's just taming out the loud peaks and keeping the volume even.

I used a fast attack and a fast release time to keep the alto sax punchy and avoid messing up the tone.

I used a ratio of **3:1** and a gain reduction of about **-3dB**, with a makeup gain of **3dB**.

Equalizer Setting For Sax

This is the trickiest part because it really depends on how the sax was recorded and what your end goal is.

So I'm not going to generalize the EQ settings; I'll show you what I did to make the saxophone I'm working on sit well in the mix.

I simply added some punch and warmth by applying a boost at 200 Hz.

The sax had some boxiness, and I had to clean that by cutting at **492hz**.

For the upper mids, I removed some nasal noise at **1.2kHz** and boosted the presence at **5.1kHz**.

Finally, I added some air by creating a high-shelf boost at **11kHz**.

Here's a saxophone EQ chart I made, and I hope it helps you properly equalize your sax sounds. I know it's not accurate, but it will guide you in the right direction.

Rumble: below 110Hz

Bottom/Punch: 125Hz – 250Hz

Fullness: 250Hz- 450Hz

Honk/Nasal: 500Hz – 1.6kHz

Presence/Edge: 2kHz – 6kHz

Definition: 6kHz – 8kHz

Air: 10kHz – 17kHz

Hiss: 17kHz

Effect and Saturation For Saxophone

For the final touches, I added reverb, delay, and tape saturation to help the sax sit well in the mix and create space, depth, and width.

The saxophone was recorded in mono, so a **mono reverb** was a good choice, and it sounded good.

A plate or room reverb works really well on woodwind and brass sounds; for this one, I chose a plate.

I also removed some high frequencies from the reverb signal and used a medium reverb time (tail).

The sound was mono, so to add some stereo, I used a stereo delay.

But it was a timed delay, meaning it was in sync with the BPM of the song.

A slap wasn't a good option because I added more width, so I used a timed ping-pong delay.

To add some harmonic excitement, I added some tape saturation to the sound.

I tried a lot of distortion plugins on this one, and the FabFilter Saturn plugin added some nice harmonics, warmth, and a vintage sound to the sax.

After all that processing, the sax mix sounded really dope.

Remember to keep the sax sounding natural and use a subtle amount of processing on it.

I had to exaggerate my settings for this tutorial so that everything would be obvious to a beginner's ear.

Here are the final results:

Mixed Saxophone:

<http://talkinmusic.com/wp-content/uploads/2014/10/Mixing-Sax-Blog-Post-Mixed.mp3>

Mixed Sax With The Music:

<http://talkinmusic.com/wp-content/uploads/2014/10/Mixing-Sax-Blog-Post-With-The-Music.mp3>

As you can hear, that was a major difference from the original source.

I hope this tutorial helps you get your sax to gel well with the entire mix and become one with the song.

How To Mix Hi-Hats And Cymbals

For our final tutorial, I want to share with you some really neat tricks for mixing hi-hats and cymbals.

When the cymbals and hi-hats are all over the place, it can be hard to get a good frequency balance for your mix.

Hats and cymbals can mess up the top end when not processed well or just ignored.

For instance, if the volume of the hi-hats and crash is too loud, they'll create masking with the vocalist's air or breath. As a result, this will prevent the vocals from shining in the mix.

Sometimes you might have thin hi-hats or ride cymbals that you want to make crisp and clean without messing up the top end.

In this tutorial, I'll share a few techniques you can use to make sure you get a good top-end balance.

Using Envelopes

For those who don't know, an "envelope" is a common way synthesizer parameters specify how a sound evolves over time.

In simple terms, this determines the length of the sound as well as how it comes in and out.

The common parameters for an envelope are attack, decay, sustain, and release.

You can use these parameters together with the filter cutoff, resonant and LFO parameters to get a good sounding shaker, tambourine, cymbal, hat etc.

... without the need for an equalizer or even compression. There are no rules for envelope shaping, but you need to create settings that work well with the other sounds in the mix.

For instance, a closed hi-hat, tambourine, or shaker will work well with a fast attack, medium decay, short sustain, and quick release.

An open hi-hat, splash, or crash will work well with a fast attack, fast decay, long sustain, and release.

Those settings might work or not, depending on the material you're working on.

But before jumping to any 3rd-party processing, play around with the synthesizer parameters, and you might not even need any further processing.

De-Essing Hi-Hats & Cymbals

Nowadays, the majority of samples and VT are processed.

This is why you need to be careful when processing sampled drums.

If your hats or cymbals are sounding harsh, a de-esser is another great tool you can use to treat them.

The great thing about using a de-esser is that it will only affect the problem frequencies without messing with the overall volume.

Especially if you're having trouble finding a good balance for the high frequencies.

There are no strict rules for de-essing hi-hats or cymbals, but I would say de-ess from the **10kHz** range going up, so that you don't mess up the presence or clarity.

Equalization

If the above techniques fail, you can use an equalizer to shape your high-frequency sounds.

If your drums were not recorded live, then I would advise you to use subtractive EQ instead of additive EQ, unless it is needed.

Equalizing hi-hats and cymbals is pretty easy.

These sounds usually have a harsh metallic sound around the **200Hz** range; some call this the "clang." To remove that, simply use a high-pass filter until around **300Hz**.

The high-pass filter will also remove unwanted mud and some mic bleed from the snare drum. If the hi-hat or cymbal is sounding thin, a boost around 400Hz to 800Hz might add the weight you want.

If the sound is too harsh, make a high-shelf cut until around **16kHz**. A small boost with a wide Q in the 3kHz range will add presence to the sound.

If the hat or cymbal is sounding dark and you need to make it bright, then make a bandpass filter boost around **9kHz** to **12kHz**, that will add some sparkle.

To add some clarity, boost around **6kHz** to **8kHz**.

Finally, a small, wide Q cut from **800Hz** to around **1.2kHz** will make room for other sounds in the mix or remove any nasal sound.

Additional Techniques

Another great technique for dealing with harsh hi-hats and cymbals is to use a multi-band compressor.

I always group the high-frequency sounds into a bus channel and add a multiband compressor to keep their volume constant throughout the entire mix.

Compress your high-frequency sounds separately from other drum sounds so that they don't keep poking into the mix or create high-frequency masking, and this also glues them together.

Use delay to create space, width, and depth for hi-hats and cymbals.

Reverb tends to create a washy sound and mess up the top end, which is why I prefer a delay effect instead.

You could also experiment with tape saturation to make the sound warmer, round off the transients, and add analogue flavor to the sound.

When it comes to panning, I pan the crash cymbal to the left and the ride cymbal to the right. Like a live drum kit, the hi-hat will be far to the left of the crash.

Final advice would be to keep your high-frequency sounds as low as possible because, as we grow older, it becomes harder to hear the top-end precisely, and then we end up exaggerating it.

You can also use headphones and reference tracks to make sure you're getting a good balance for the top-end of your mix.

That's all, my friend; thank you for sticking with me this far.

If you need some help, don't hesitate to contact me, and many thanks for investing in my course; I hope it helps you get the sound you desire.

2 TIPS FOR FINALIZING THE MIX

At this point, you already know what it takes to transform a home studio demo song and turn it into a professional song that is ready for the masses.

In this section of the guide, we'll take the knowledge you already have and take it to the next level.

But before you can do that, please make sure that you test the strategies mentioned above before jumping into the advanced stuff, because there won't be a lot of explaining here; it's just tactics.

If you're still wondering how loud the vocals should be, then this section is not for you. Refer to the previous lessons before reading this section.

Since you already know how important mix bus compression is, we'll look at other bus compression techniques.

Advanced Buss/Group Compression

It is critical to group your sounds and compress them from the group channels because it glues all of your sounds together, making them sound like one.

Another neat trick is to group the sounds in your mix based on which frequency they dominate. So you'll have three group channels called bass, midrange, and treble.

Basically, you'll take your bass, kick, and floor toms to the bass group channel. Next, you group the male vocals, electric guitar, piano, and snare, then group them in the midrange group channel.

Finally, you'll take the hi-hats, tambourine, overheads, shaker, acoustic guitar, and female vocals to the treble group channel.

Compress these groups and adjust levels to taste.

Just don't get carried away and over-compress because you still have the mix bus compressor and you've compressed individual sounds that you sent to groups based on instruments like I taught you above.

This technique is just to create glue and make sure that you get a good overall balance.

Digital Analogue Summing

If you have an analogue mixer, then route all the tracks to the mixer to create analogue summing and give your mix warmth and depth.

Or you can replicate that by using NLS Non-Linear Summer by Waves Audio.

That's what I use to finalize all my mixes to make them sound like they were engineered with an SSL (solid state console), a Neve (vintage British console), or the classic console owned by Mike Hedges.

The NLS Non-Linear Summer gives you three legendary consoles in one powerful plugin.

It also includes some really nice sounding harmonics that are emulated from three different analogue consoles and their bus channels to assist you in achieving analogue sound in the box.

It's pretty straight-forward, and you can read the manual to know how it works.

Another great company that offers analog summing in the box is Slate Digital.

CONCLUSION

If you're reading this part of the book, then I would like to take this moment to thank you not only for downloading the book but also for reading everything to the end.

I have a glossary below that will really help you understand any terms that might have been confusing while reading the guide.

The way you approach a mix is entirely about what you feel comfortable with.

There is no right or wrong way to do things. You can mix while you are busy making the music or have a mixing session.

Mixing while making the music drains my energy to finish the song, so I do the sound selection and mixing later, just so I can finish putting the idea in the software while it's still fresh.

When I have the whole idea, I start choosing sounds that go well together.

As I'm choosing the sounds, in my mind I already have a picture of how the final mix is going to sound.

Because I've been making the song for maybe 6 weeks or even more, by the time I open the software to do the mixdown, I already have a rough idea of the EQ, compressor, reverb, and delay settings for all my sounds.

That's how I do my mixes.

If it's someone's mix, then they have to give me an idea of how they want everything to sound or send me their rough mixdown.

Being a mixing engineer can be a really great career, but if you're reading this book because you want to mix your own music, then I hope this information does help you reach your goal and make your music sound like your favorite songs.

Read this again as many times as you can to really understand it; going through it once won't help.

That's it from me, and now it's all up to you to put all this information into practice.

Warmest Regards,

Gugulethu

support@audiospectra.net

GLOSSARY

Credits for glossary: www.testing1212.co.uk

ADSR:

The letters A, D, S & R are the first letters of: Attack, Decay, Sustain and Release. These are the various elements of volume changes in the sounding of a keyboard instrument.

Amp

- 1) An abbreviation of the term Amplifier (A device which increases the level of an electrical signal).
- 2) An abbreviation of Ampere (the unit of current).
- 3) An abbreviation of amplitude (the height of a waveform above or below the zero line).

Amplification

An increasing of signal strength.

Analog (Analogue)

Representative, continuous changes that relate to another quantity that has a continuous change.

Attack

The rate the sound begins and increases in volume.

Audio

Most often referring to electrical signals resulting from the sound pressure wave being converted into electrical energy.

Automation

In consoles, a feature that lets the engineer program control changes (such as fader level) so that upon playback of the multitrack recording these changes happen automatically.

Aux Send

Short for the term Auxiliary Send (a control to adjust the level of the signal sent from the console input channel to the auxiliary equipment through the aux buss).

Balance

- 1) The relative level of two or more instruments in a mix, or the relative level of audio signals in the channels of a stereo recording.
- 2) To make the relative levels of audio signals in the channels of a stereo recording even.

Balanced

- 1) Having a pleasing amount of low frequencies compared to mid-range frequencies and high frequencies.
- 2) Having a pleasing mixture of the various instrument levels in an audio recording.
- 3) Having a fairly equal level in each of the stereo channels.
- 4) A method of interconnecting electronic gear using three-conductor cables.

Bandwidth

- 1) The range of frequencies over which a tape recorder, amplifier or other audio device is useful.
- 2) The range of frequencies affected by an equalization setting.

Bar

A term meaning the same thing as the term Measure (the grouping of a number of beats in music, most-often four beats).

Bass

- 1) The lower range of audio frequencies up to approximately 250 Hz.
- 2) Short for Bass Guitar.
- 3) Lower end of the musical scale. In acoustics, the range (below about 200 Hz) in which there are difficulties, principally in the reproduction of sound, due to the large wavelengths involved.
- 4) The lower frequencies.
- 5) On the soundboard this should refer to the bass guitar channel, not the bass drum.
- 6) The lowest frequencies of sound. Bi-Amplification uses an electronic crossover or line-level amplifiers for the high and low frequency loudspeaker drivers.

Blending

- 1) A condition where two signals mix together to form one sound or to give the sound of one sound source or one performance.
- 2) Mixing the left and right signal together slightly which makes the instruments sound closer to the center of the performance stage.
- 3) A method of panning during mixing where instruments are not panned extremely left or right.

Boost

To increase gain, especially to increase gain at specific frequencies with an equalizer.

Bottom

The bass frequencies (as in "needs more bottom end").

Boomy: a build-up of low frequencies—often in low-pitched drums—that causes an overpowering emphasis on the sustain of the sound

Boxy: a lack of low and high frequencies; a sound that has too much midrange

Buss (Bus)

A wire carrying signals to some place, usually fed from several sources.

Cancellation

A shortening of the term Phase Cancellation (the energy of one waveform significantly decreasing the energy of another waveform because of phase relationships at or close to 180 degrees).

Channels

These are divided into two separate categories. Input channels are those channels coming into the soundboard such as microphones and direct lines. Output channels are those leaving the board such as monitor and main outputs.

Chorus

Common type of effect that makes sounds fuller and thicker.

Clean

Describes a distortion free sound with few effects.

Clipping

Distortion of a signal by its being chopped off. An overload problem caused by pushing an amplifier beyond its capabilities. The flat-topped signal has high levels of harmonic distortion

which creates heat in a loudspeaker and is the major cause of loudspeaker component failure.

Compressor

- 1) Effect used to squash the sound together. Used properly, it can take the edge off or your sound. Used improperly, it can take the life right out of your system and make it sound like an MTV mix.
- 2) A piece of sound processing equipment that ensures all wanted signals are suitably placed between the noise and distortion levels of the recording medium. It evens out the unwanted changes in volume you get with close-miking, and in doing so, adds punch to the sound mix. A Limiter is used to stop a signal from exceeding a preset limit. Beyond this limit, the signal level will not increase, no matter how loud the input becomes. A Limiter is often used to protect speaker systems (and human ears) by preventing a system from becoming too loud.

Crisp

Describes a good clean high midrange sound. It can be good or bad depending on the look on the face of the guy who said it.

Crunchy

Slightly distorted as a result of over-compression, over-limiting, clipping, or intentional overdrive.

Comb Filter

- 1) The frequency response achieved by mixing a direct signal with a delayed signal of equal strength especially at short delays.
- 2) Loosely used to also describe effects that can be achieved with comb filtering as part of the processing.

CPU

Abbreviation of Central Processing Unit (The main “brain” chip of

a computer or the main housing of a computer that contains the "brain" chip).

Cut

- 1) One selection (one song) on a pre4ecorded music format.
- 2) A term with the same meaning as Mute (to turn off a channel or a signal).
- 3) To reduce gain of a particular band of frequencies (with an equalizer).
- 4) To not pass a particular band of frequencies (said of a filter)

DAW

A digital audio workstation is an electronic device or application software used for recording, editing and producing audio files.

Decibel (dB)

- 1) Relative measurement for the volume (loudness) of sound. Also used to measure the difference between two voltages, or two currents. See Zero dB.
- 2) A numerical expression of the relative loudness of a sound. The difference in decibels between two sounds is ten times the common logarithm of the ratio of their power levels.

Delay (Digital, Analogue)

Effect used to create echo...echo...echo...echo...echo...

Depth

Back to front in the stereo image. Differentiation between close and distant sounds.

Dry

Describes a sound coming from the PA with no effects on it.

Early Reflections

The first echoes in a room, caused by the sound from the sound source reflecting off one surface before reaching the listener.

Effects

- 1) Various ways an audio signal can be modified by adding something to the signal to change the sound.
- 2) Short for the term Sound Effects

Envelope

- 1) How a sound or audio signal varies in intensity over a time span.
- 2) How a control voltage varies in level over time controlling a parameter of something other than gain or audio level.

Equalizer (Parametric, Graphic)

This is used to filter out and adjust specific frequencies in the PA. This is the part of the PA where you have the most control over the band's overall sound. It is also the number one weapon against feedback.

Expansion

The opposite of compression; for example, an expander may allow the signal to increase 2 dB every time the signal input increased by 1 dB.

Fade

A gradual reduction of the level of the audio signal.

Fader

A control to control the gain of a channel on the console, thereby determining the level of the signal in that channel.

Fat (PHAT)

Having more than a normal amount of signal strength at low frequencies or having more sound than normal.

Filter

A device that removes signals with frequencies above or below a certain point called the cut-off frequency.

Final Mix

The two track stereo master tape which was mixed from the multitrack master.

Frequency

The number of cycles of a waveform occurring in a second.

Frequency Range

The range of frequencies over which an electronic device is useful or over which a sound source will put out substantial energy.

Fx

Short for effect.

Gain

- 1) Knob usually found at the top of each input channel on the soundboard. Used to set input levels of the separate channels to relatively equal positions.
- 2) The amount of increase in audio signal strength, often expressed in dB.

Gain Reduction

The working of a limiter or compressor reducing gain during high-level passages.

Gate

A dynamic processing device that turns a channel off or down when the signal drops below a certain level.

Group

A number of channels or faders that can be controlled by one Master VCA slide.

Grouping

Controlling the gain of several individual channels with a Group Fader.

Harshness

An excessive amount of high frequencies

Haas Effect

Simply stated, a factor in human hearing where delay has a much bigger effect on human perception of direction than level does.

Harmonics

Integer multiples of a fundamental frequency, the fundamental itself being the first harmonic, its first overtone the second harmonic, etc. Attributing to instruments, voices, etc. their distinctive timbre.

Headroom

The level difference (in dB) between normal operating level and clipping level in an amplifier or audio device.

Hertz

The unit of frequency. Equivalent to cycles per second.
Abbreviation: Hz.

kHz

An Abbreviation of kilo-Hertz.

Layering

The recording (or playing) of a musical part with of several similar sound patches playing simultaneous.

Lead

The musical instrument that plays the melody of the tune, including the vocal.

Level

The amount of signal strength; the amplitude, especially the average amplitude.

Limiter

A device which reduces gain when the input voltage exceeds a certain level.

Masking

The characteristic of hearing by which loud sounds prevent the ear from hearing softer sounds of similar frequency.

Master Fader (Channel)

The fader which controls the main output(s) of the console during mixdown.

MIDI

Short for Musical Instrument Digital Interface; a digital signal system (a system of number signals) used to communicate performance information to and from musical instruments making music.

Mix

- 1) To blend audio signals together into a composite signal.
- 2) The signal made by blending individual signals together.
- 3) A control or function on a delay effects/reverberation device which controls the amount of direct signal that will be mixed into the processed signal.

Mixer

A console, or other device that blends audio signals into composite signals and has a small number of outputs.

Mixing Solo

A button that turns off all other channels, allowing the signal to be heard in the stereo perspective and level used in the mixdown, and with the reverberation being used.

Mono

Shortened from Monophonic and meaning that there is only one sound source or the signal was derived from one sound source.

Muddy

Describes a low end muffled sound lacking highs and mids, and possibly having too much effects.

Mute Switch

A switch which turns off a channel, takes out a track signal from the monitors, or which turns off the entire monitor signal.

Noise

Any unintentional or objectionable signal added to an audio signal.

Octave

A difference of pitch where one tone has a frequency that is double or one-half of the frequency of another tone.

Pan (Balance)

Knob on the mixer that adjusts the relative volume between left and right (or A and B) in a stereo setup.

Parallel

Blending an unprocessed (Dry) signal with processed (Wet) signal.

Parameter

Each adjustment that is possible to change in a device.

Peak

The highest point in the audio waveform.

Phase

- 1) The amount by which one sine wave leads or lags a second wave of the same frequency. The difference is described by the term phase angle. Sine waves in phase reinforce each other; those out of phase cancel.
- 2) A measurement (expressed in degrees) of the time difference between two similar waveforms.

Phase Cancellation

The energy of one waveform decreasing the energy of another waveform because of phase relationships at or close to 180 degrees.

Pre Fader

A placement of a send control (or other control) before the main channel fader.

Presence Frequencies

The range of audio frequencies between 4 kHz and 6 kHz that often, when boosted, increases the sense of presence, especially on voices.

Preset

A program of a sound done at the factory by the manufacturer.

Pumping Effect

The sound of the noise changing volume as the limiter or compressor works.

Q

The sharpness of the peak response in an equalization circuit.

Release

The rate that the volume of a synthesizer drops to no-sound once the key is released.

Resonance

The prolonging of the sound at a certain frequency and the tendency of something to vibrate at a particular frequency after the source of energy is removed.

Roll-Off

The reduction of signal level as the frequency of the signal moves away from the cut-off frequency, especially when the cut-off rate is mild.

RMS

Abbreviation for root mean square. The effective value of a given waveform is its RMS value. Acoustic power is proportional to the square of the RMS sound pressure.

Rumble

A low-frequency noise, especially that caused by earth/floor vibration or by uneven surfaces in the drive mechanism of a recorder or playback unit.

Send

A control and buss to feed signals from the console channels to some outboard device such as a reverberation effects unit.

Sibilance

Energy from a voice centred around 7 kHz caused by pronouncing "s", "sh" or "ch" sounds.

Stereo

A recording or reproduction of at least two channels where positioning of instrument sounds left to right can be perceived.

Stereo Image

The perception of the different sound sources being far left, far right or any place in between.

Transient

The initial high-energy peak at the beginning of a waveform, such as one caused by the percussive action of a pick or hammer hitting the string, etc.

Tempo

The rate at which the music moves measured in Beats Per Minute.

Timbre

The timbre of the instrument is what makes an instrument sound like that instrument and not another, even though the other instrument may be playing the same pitch.

Tone

- 1) One of several single-frequency signals at the beginning of a tape reel at the magnetic reference level that will be used to record the program.
- 2) Any single-frequency signal or sound.
- 3) The sound quality of an instrument's sound relative to the amount of energy present at different frequencies.
- 4) In some synthesizers, a term meaning the audio signal that will be put out by the unit which would be similar to the sound of an instrument.

Tonal balance

The distribution of energy across the audio spectrum

Tweak

A slang term for calibration (a setting of all operating controls and adjustments for optimum performance of a device) especially very precise calibration.

Warmth: a tonal quality characterized by mild levels of even harmonic distortion

Waveform

The shape made by the fluctuations of a quantity over time.

Wet

Having reverberation, ambience, delay or any other audio effect.

Width

How wide the stereo channel should be from left to right. The perceived difference in left-to-right spacing between signals.