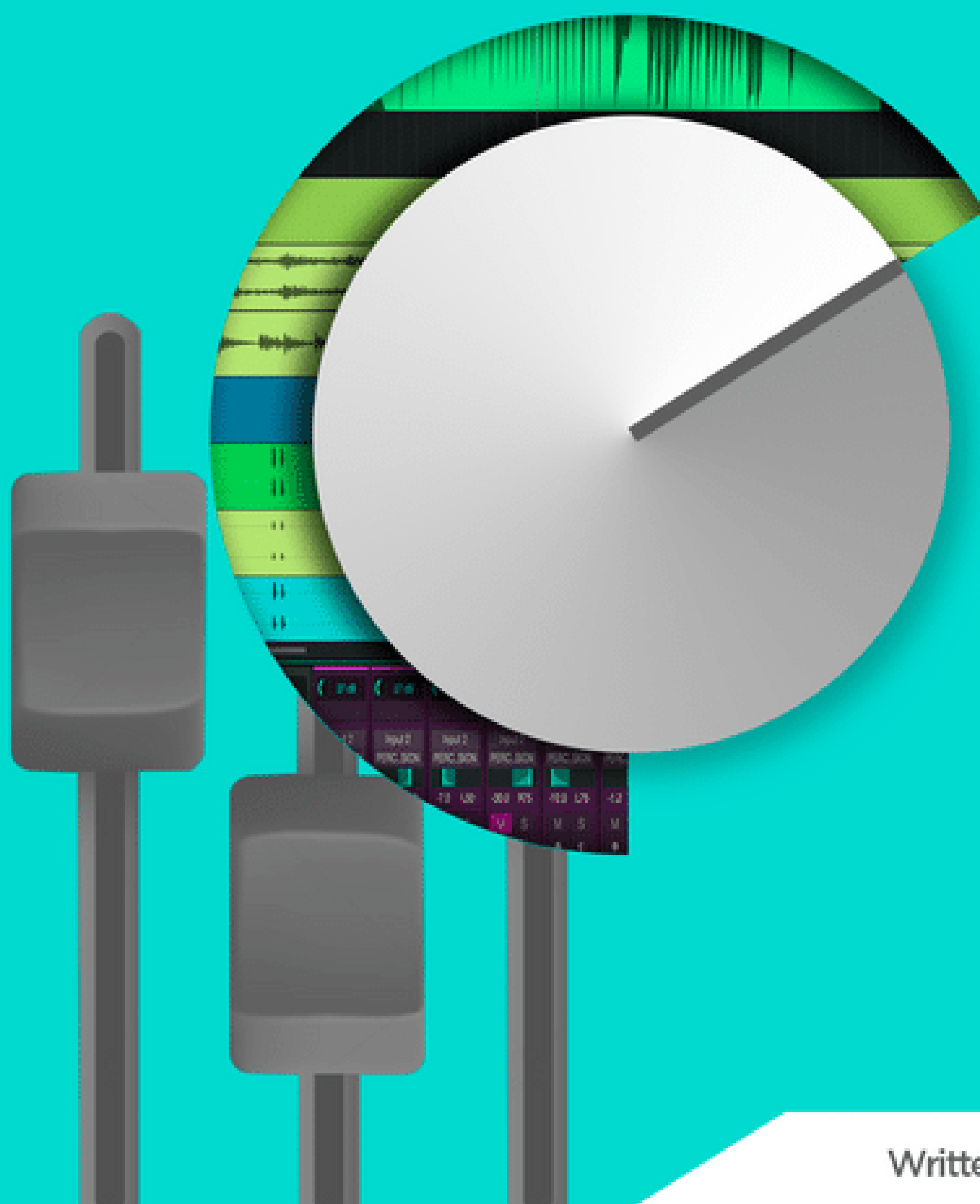


# MIXOLOGY

A COMPLETE GUIDE FOR  
PROFESSIONAL MIXING



Written by Jay Dwivedi

Create professional mixes  
**EVERYTIME** you hit mixing sessions

# MIXOLOGY

## A COMPLETE GUIDE FOR PROFESSIONAL MIXING

Written by  
JAY DWIVEDI

Designed by  
JAY DWIVEDI

Presented by  
MIDISIC

Create professional mixes  
**EVERYTIME** you hit mixing sessions

Copyright © 2021 midisic.

All rights reserved. No part of this publication may be reproduced, distributed, or transmitted in any form or by any means, including photocopying, recording, or other electronic or mechanical methods, without the prior written permission of the publisher, except in the case of brief quotations embodied in critical reviews and certain other noncommercial uses permitted by copyright law. For permission requests, write to the publisher, at the address below.

info@midisic.com (Publisher)

jay@midisic.com (Author)

Any references to historical events, real people, or real places are used fictitiously. Names, characters, and places are products of the author's imagination.

<https://midisic.com>

# STREET TWEET TUNING BOOK

**01** *INTRODUCTION*

**02** *PROBLEMS IN YOUR STUDIO*

**03** *OPTIMIZING LISTENING POSITION*

**04** *ACOUSTIC TREATMENT OF STUDIO*

**05** *PREPARING FOR MIX*

**06** *THE ART OF MIXING*

**07** *MIXING ESSENTIALS*

**08** *PANNING*

**09** *EQUILISATION*

**10** *COMPRESSION*

**11** *REVERB*

**12** *DELAY*

**13** *SATURATION*

**14** *MONITORING YOUR MIX*

**15** *REVISION*





# INTRO DUCTION

# 01

Hey everyone

First of all, a big thank you to everyone who supported our work by buying Mixology, and now it's my time to serve you all the knowledge I got by working on various projects that I mixed and mastered over years.

I believe that this is a must-have ebook for anybody making music and who wants to achieve a professional mix which they can feel proud of. It's not defined by any genre or DAW. I've designed this ebook so that you will get the most out of it as we have focused on techniques that you should use to get the best out of your mixes and not the software or hardware that you should have. I recommend you to read the whole book thoroughly so you don't miss any part. It's time for you to learn the essentials of mixing your music. Let's begin.

So without further ado let's get started.....

**So what does it mean to mix your music well in excess essence i.e.**

Mixing music is actually very straightforward. The overall goal is to create a good balance of the overall sounds used in the project. You do this by mixing all individual tracks, using various techniques to call out their own unique space. In the final mix, you are striving to get a good degree off-separation between every single track in your project so that they each have their own place, their own domain in the mix.

Yes, tracks will overlap with each other. That's natural because total separation is basically impossible. But when you add up all the techniques you will learn in this ebook, you will be able to create better separation between tracks, which will lead to higher clarity for all instruments and sounds in your mix.

Imagine a lumpy restaurant full of people talking all around you. The overall sound will be like a big roll-off noise. You will be able to hear the noise created by the people, but not individual conversations and this creates a messy sound that nobody loves.

This analogy is like a bad mix in music production, with lots of instruments and tracks fighting with each other, which will create a big, muddy mess of sound. But let's say you had volume control for each person in that restaurant. Let's say you could pan them differently in the stereo space, EQ, and filter their voices or automate their volume. Use compression to tame the audio balance and so on.

Then you'll be able to separate them to hear what each person is talking about. That's an analogy for what good mixing is about.

So here I am to teach you everything from beginning to end. So hold on tight and let's move further with the topic.

## **HOW WE WILL LEARN MIXING?**

So, the first thing I want to talk about is how to learn mixing and how I'm going to guide you through this ebook.

Now imagine mixing as an instrument, you can't just simply learn to play guitar or piano by just watching a tutorial, right? You have to practice it again and again. It takes a lot of experimentation and gradually

you will be able to master it. Be it anything, instruments or mixing, you have to keep on practicing everything that you'll learn. And here we have provided you with some of the best materials that will help you speed up your learning process. You just have to take the lessons that I'm providing and put those into your arsenal of tricks. So when you approach a piece of music that you're trying to mix, you will have all the tools that you can use.

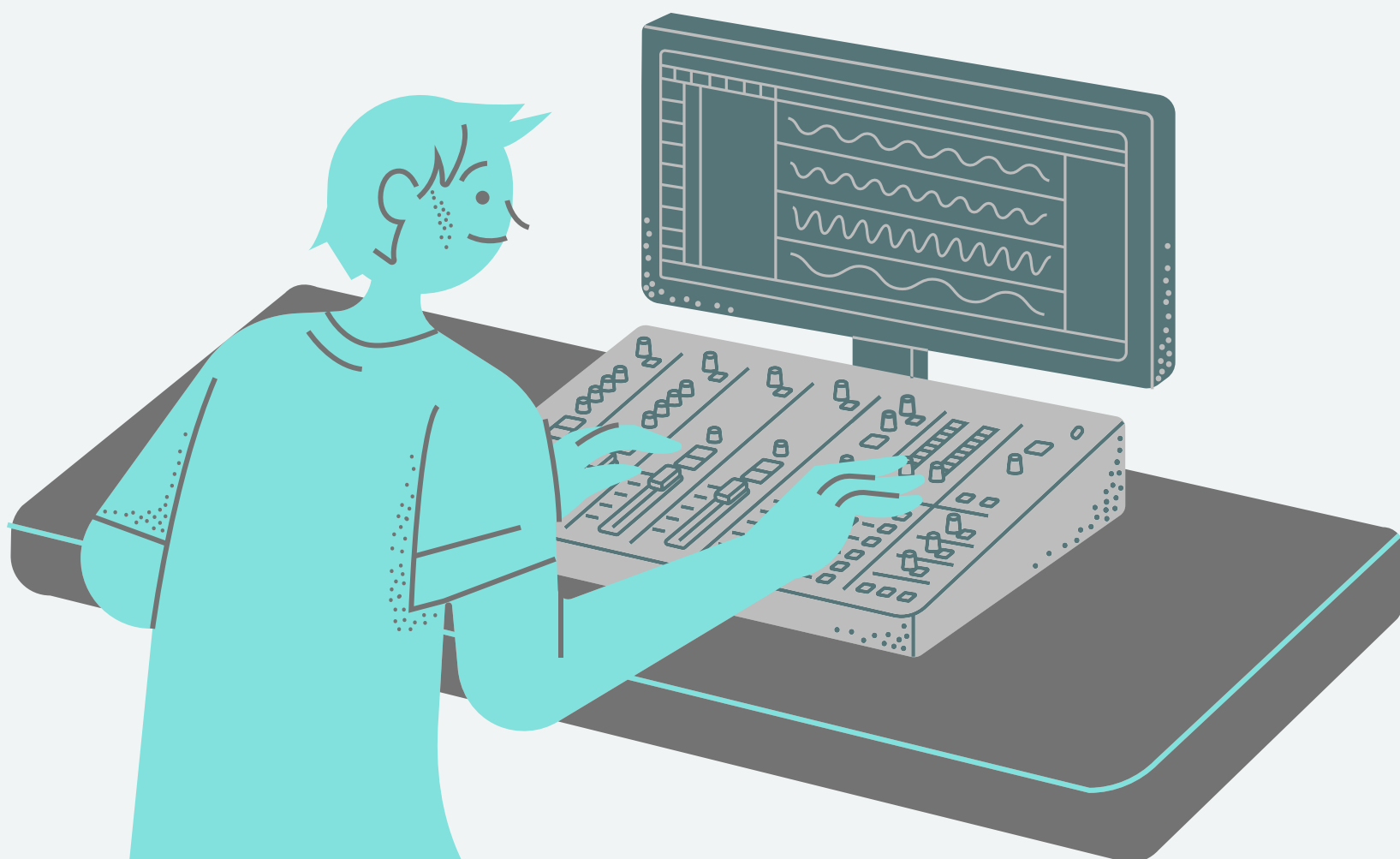
I want you to consider this book as an everyday class, and not as any book that you have to finish reading. So that you can practice it side by side on your own tracks. Actually, I would suggest you to work on tracks that are not yours. Practicing on other people's tracks will be really helpful for you. If you only work on your tracks, you're gonna learn how to mix your music well, but not necessarily be a person who mixes all music really well.

So try to work on your own music, but also keep in mind to mix some music produced by your friends. You can also try mixing music of different genres, basically anything you can get your hands on but don't expect to sound amazing right away. The more you do it, the better you'll get at it

So keep a thing in mind throughout this whole guide that mixing is something that takes practice, and if you keep working at it and you use the techniques that I'm going to show you, you will get better at it.

Okay, now, the last thing I want to say here is that in further we're going to start off with a bunch of basic stuff. So the first good lesson of Mixology is going to be about mixing styles, learning environment, and how to set up your sessions, things that aren't difficult but play a major role in speeding up the mixing process. These are all extremely valuable stuff, so you have to pay attention to what we will be doing before starting the actual mixing process. But before that let's understand...

## WHAT IS MIXING





Mixing is the process of taking recorded tracks, sounds, and instruments and blending them together. Tracks are blended using various processes such as EQ, Compression, Reverb, and Delay. The aim is to sculpt your arrangement to make sense of all your tracks in relation to each other.

## HOW LISTENING ENVIRONMENT/ GEARS AFFECT OUR MIX



You may think how your listening environment (home, studio or any place where you will be mixing your music) or mixing gear (studio monitors, headphones, speakers, etc.) will affect your mix.

Let me tell you that when it comes to mixing your piece of music you must listen to the 'TRUE' sounds that you have in your project.

When you mix your project with normal speakers

that have too much of bass, you probably gonna tame the low end. Now when you listen to that mix on other devices you will notice that your music is missing the low end.

The same happens with your listening environment. You may not notice, but your room enhances certain frequencies from the sound coming from speakers that you are using for mixing. So when you mix your project in a room without acoustic treatment, you will be listening to false sounds with all that reverb, echos that are produced by your room that is not acoustically treated, and hence you will not be able to mix your project perfectly.

To avoid this, you can use 'Studio Monitoring Headphones'. These headphones are built primarily for recording, mixing and mastering, because they have a flat sound and are not bass boosted like consumer headphones. Studio monitors and Monitoring Headphones try to keep the audio as authentic as possible.

So if your room is not acoustically treated you can go with Studio Monitoring Headphones, but just think of wearing a headphone for 3-4 hours for



A good pair of studio monitors can get your job done, but for studio monitors, you need to have your room acoustically treated to get the most authentic sounds out of them. So now we will learn how to do acoustic treatment of your studio or the room where you are or will be mixing your tracks.



# PROBLEMS

IN YOUR

# STUDIO

02

Acoustic treatment is one of the most important parts of your studio. An untreated room will add color to sound produced by your speakers in various ways, which means the mixing and mastering decisions you make may not be correct. This will cause your mix to not sound good on other playback systems. So let's understand how to do acoustic treatment of your studio

First, we will look at common acoustic problems that your untreated studio has that can make your mix suffer.

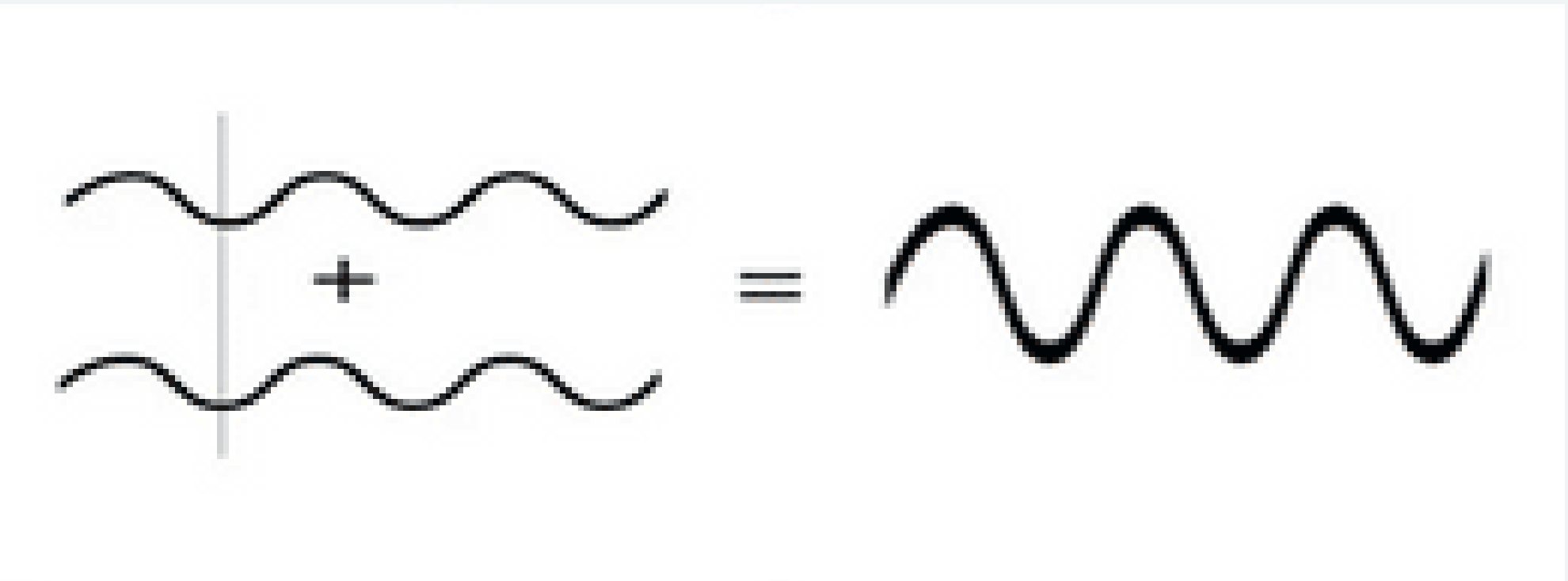
### **1. COMB FILTERING**

When a direct sound coming from monitors is combined with sound reflected by the walls of your studio something known as the comb filtering effect is created. To know why comb filtering occurs, you have to understand how sound waves work.

When two waves interfere there are 3 types of interference:

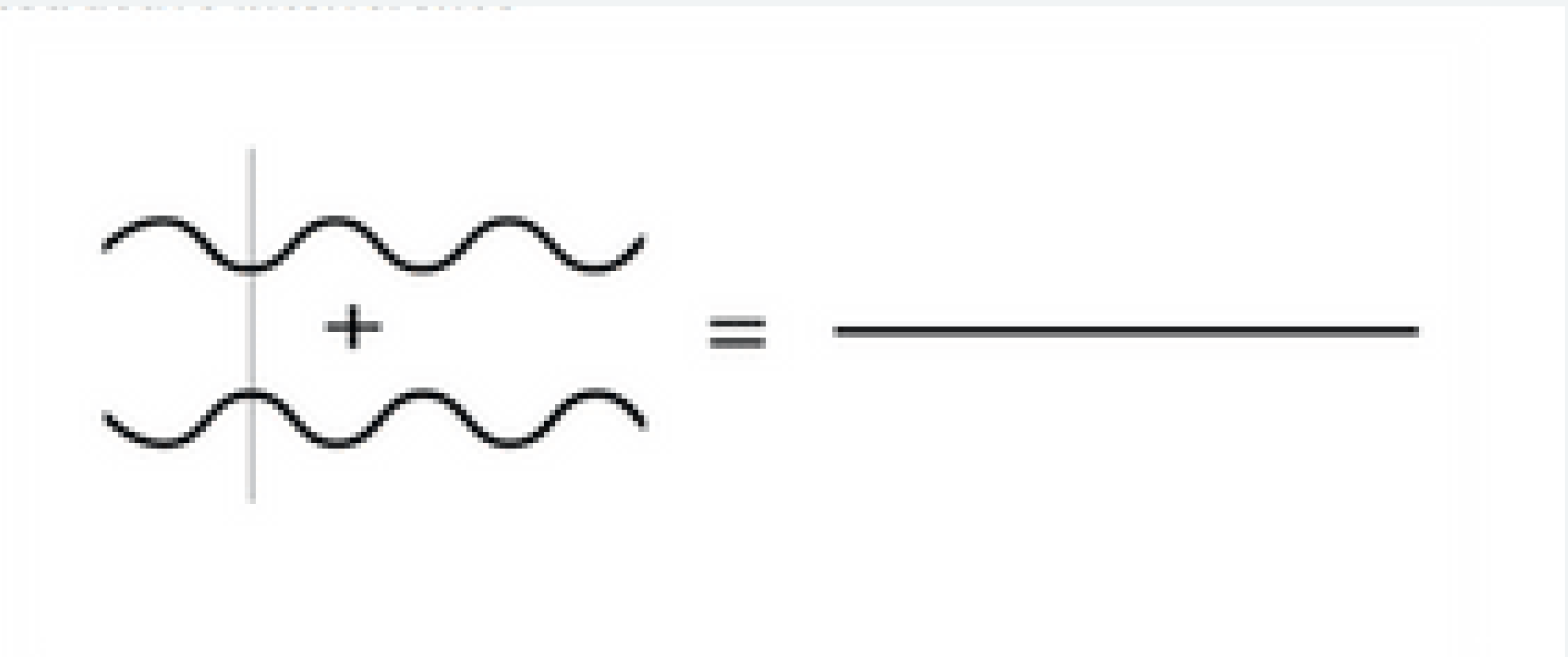
#### **A. CONSTRUCTIVE INTERFERENCE**

Take a look at these sound waves. The waves are the exact same frequency. They are also in phase,



meaning the waveforms go up and down together. Because they are in phase and have the same amplitude, they add together and create a sound wave that is double the original size. It's like  $1 + 1 = 2$ . This is called constructive phase interference.

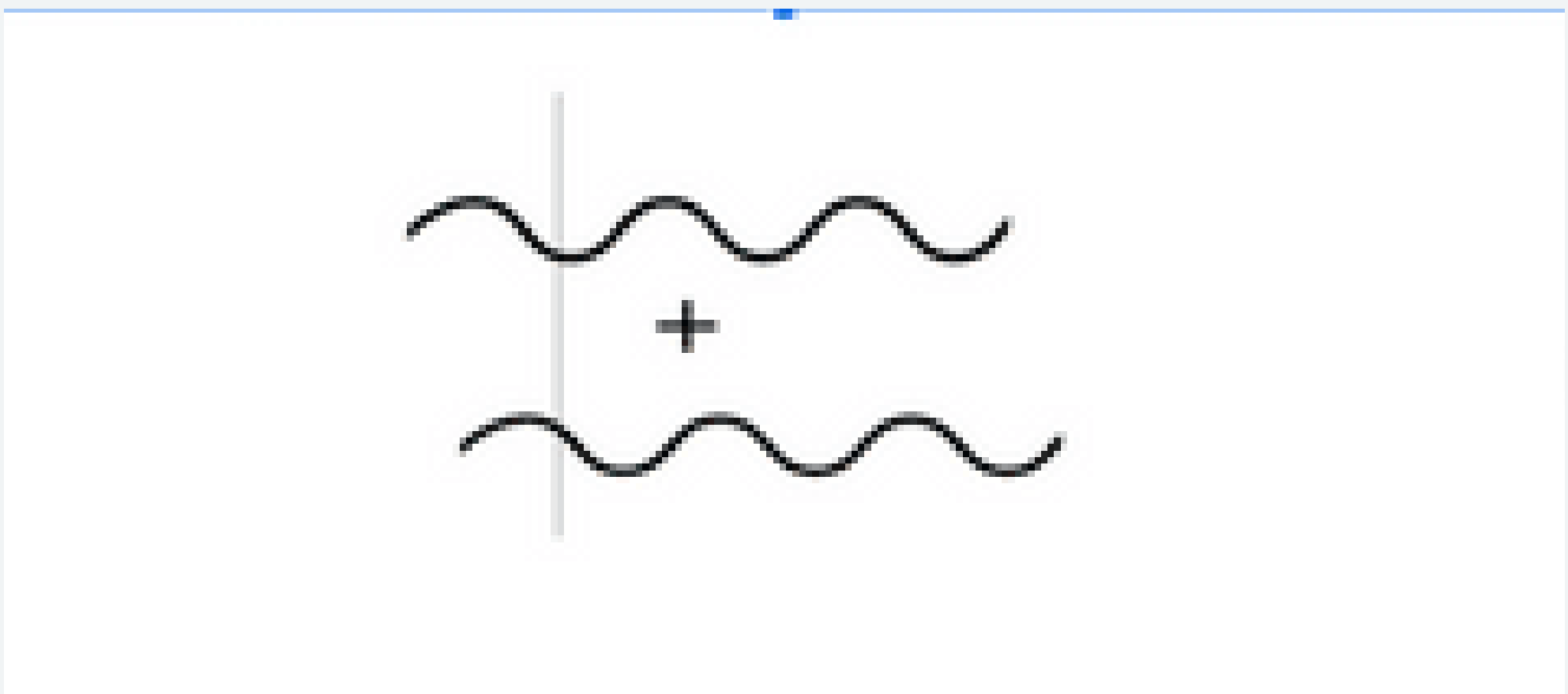
## B. DESTRUCTIVE INTERFERENCE



In this image, the sound waves are perfectly out of phase. This means that when one waveform goes up, the other goes down. The waves are the same size and perfectly out of phase, so the result is complete

cancellation. These two sounds add together to equal nothing. It's like  $1 + -1 = 0$ . This is called destructive phase interference.

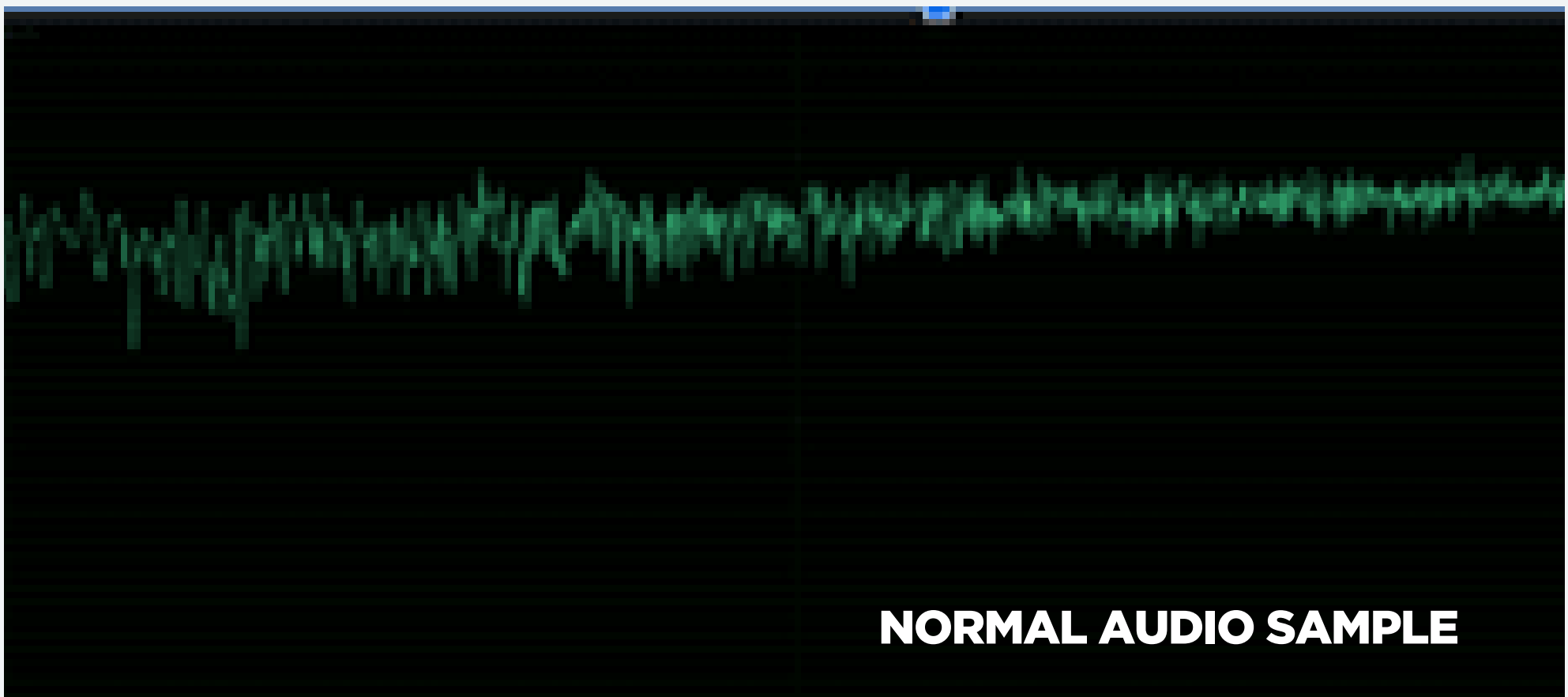
It's rare that the two waves are ever lined up perfectly in phase or perfectly out of phase. Usually, they are slightly offset as you can see below. In most cases, there is a combination of constructive and destructive interference.



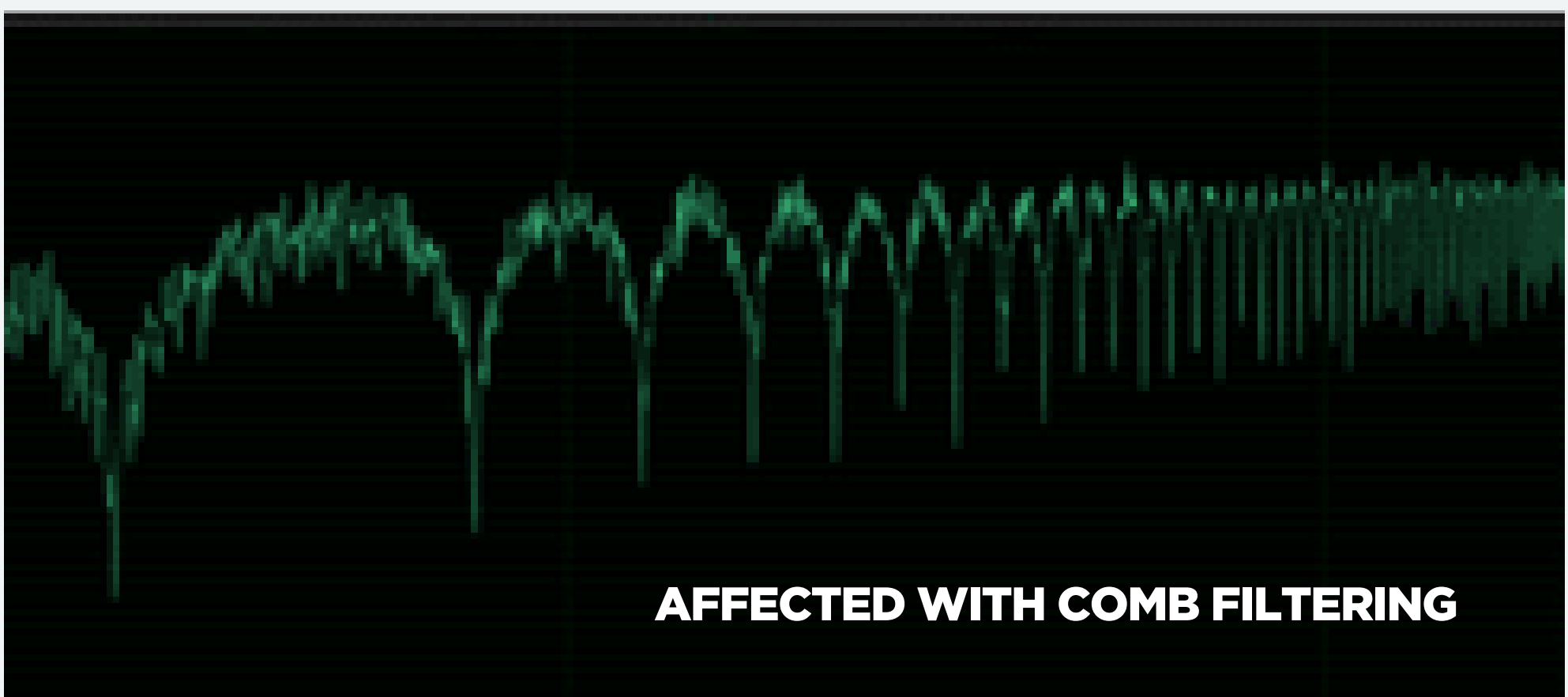
This phenomenon causes comb filtering,

Comb filtering gets its name from the shape it creates on a frequency response graph.

Just take a look at the Frequency graph of a normal sample.



Now the next is a frequency response graph of the same white noise but duplicated. The duplicated white noise has been shifted in time by 1ms. The line on this graph resembles a comb.



The comb shape forms because some frequencies are perfectly in phase and some frequencies are perfectly out of phase.

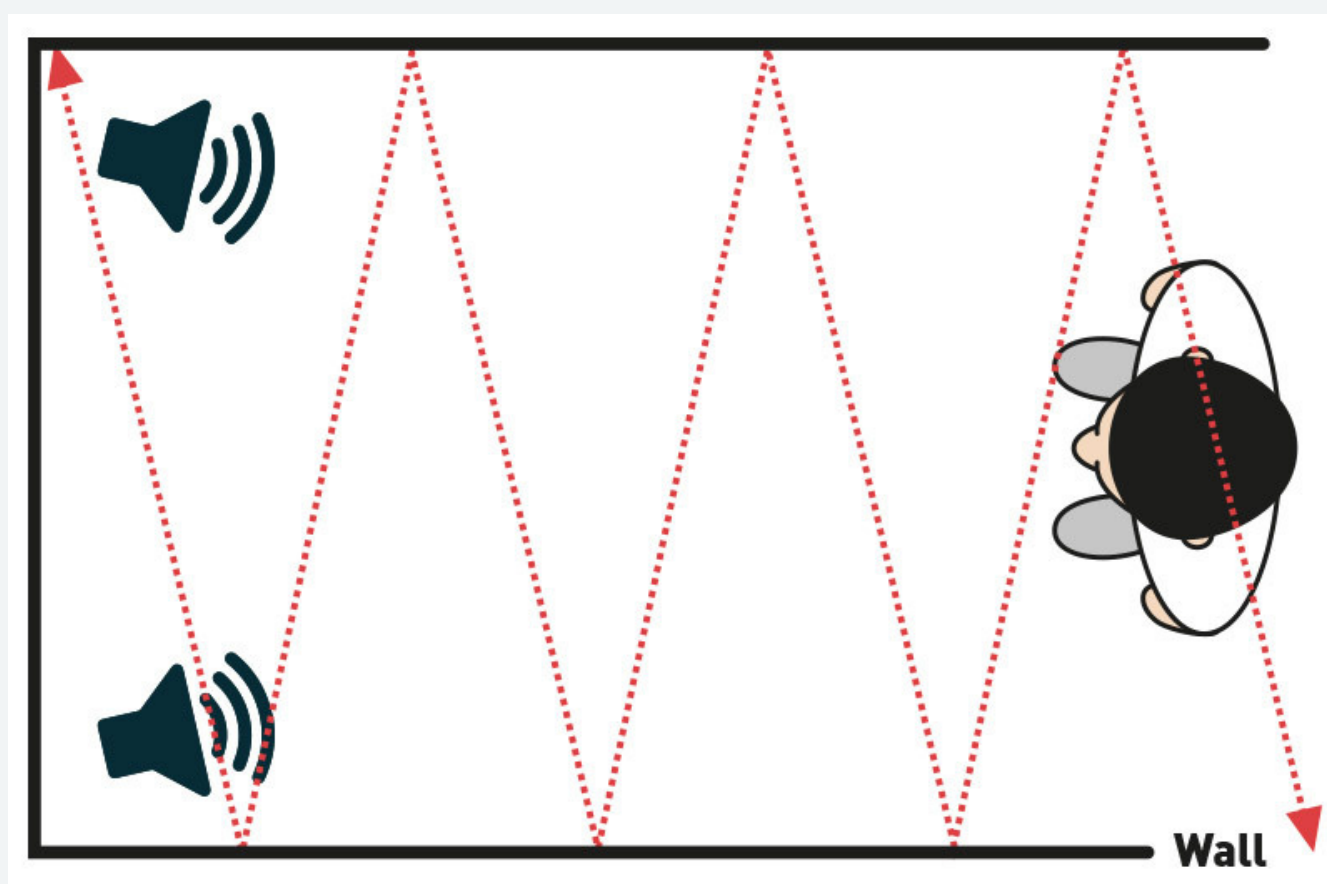
The highest points of the comb are frequencies that are perfectly in phase and sum together.

The lowest points of the comb are frequencies that

are perfectly out of phase and completely cancel. The frequencies in between are partially out of phase.

This might be too much then you need to know so all I want to say is this will ruin your mix and if you don't want this to happen you will have to acoustically treat your studio by installing bass traps, sound diffusers, etc. We will let you know more about this in a moment but first let's quickly see what other factors affect your mix, since we've only discussed one till now.

## 2. FLUTTER ECHO



An acoustic problem known as flutter echo may occur when a sound reflects back and forth between untreated parallel walls. If the time between reflections is large enough, your ears will perceive



these reflections as an echo, rather than sound that's diffusing throughout the room. This effect is strengthened by the regularity of the echoes, making it potentially quite audible to the human ear. The echo will eventually fade away due to the natural absorptive properties of the walls, but flutter echo can be detrimental to the effectiveness of a critical listening environment.

### **3. ROOM MODES, STANDING WAVES, NODES & ANTI-NODES**

The sound pressure level of any room varies at different positions and so their response towards certain frequencies of sound.

Due to this certain spots of the room, the room will be more resonant towards a certain frequency and these resonances are referred to as Room Modes.

Room modes, standing waves, nodes & anti-nodes are the main reasons you face difficulties in setting up the low-end of your music. If you've set up your mixing position at a standing wave's node, you'll hear a lack of bass, and if you've set up your mixing position at a standing wave's anti-node, you'll hear an abundance of bass.



There are still more reasons why mixing on studio monitors without acoustic treatment is not an ideal choice, but I think this may be sufficient to convince you and now we will learn about how to do acoustic treatment of your studio or home studio.

We will set up the acoustics in such a way that your studio should not be ‘too dead’ or ‘too live’.

What does it mean for a room to be live or dead?  
Let me explain:

Imagine what it sounds like when you yell in a gymnasium. The sound which you’ll hear back will be kind of echoed or reverbed. You’ll notice a clear “tail” of your yelling which decays slowly over time. This sound feels “Lively” and hence the place is referred to as a live place. (There is no official statement that states a place can be referred to as live or dead, it’s just to make you understand).

Now Imagine yelling in a car with windows rolled up. The sound decays quickly and has a “dead” characteristic.

In short, a “live” room will allow the sound to

interact with the walls, ceiling, floor causing some problems that we discussed above, while a “dead” room will almost absorb all the sound coming from the speakers, and we don’t want any of them. We have to achieve a perfect balance of those ‘live” and “dead” characteristics in our studio, not more not less and acoustic treatment will help us to achieve this.

But first, we will know how to optimize the listening position of your studio.



# OPTIMIZING LISTENING POSITION

03

Setting up the correct listening position will help you to minimize and control acoustic problems, making the application of acoustic treatment easier and potentially more affordable.

## **SETTING UP THE DESK**

Take care of the following while setting up the positions of your desk components:

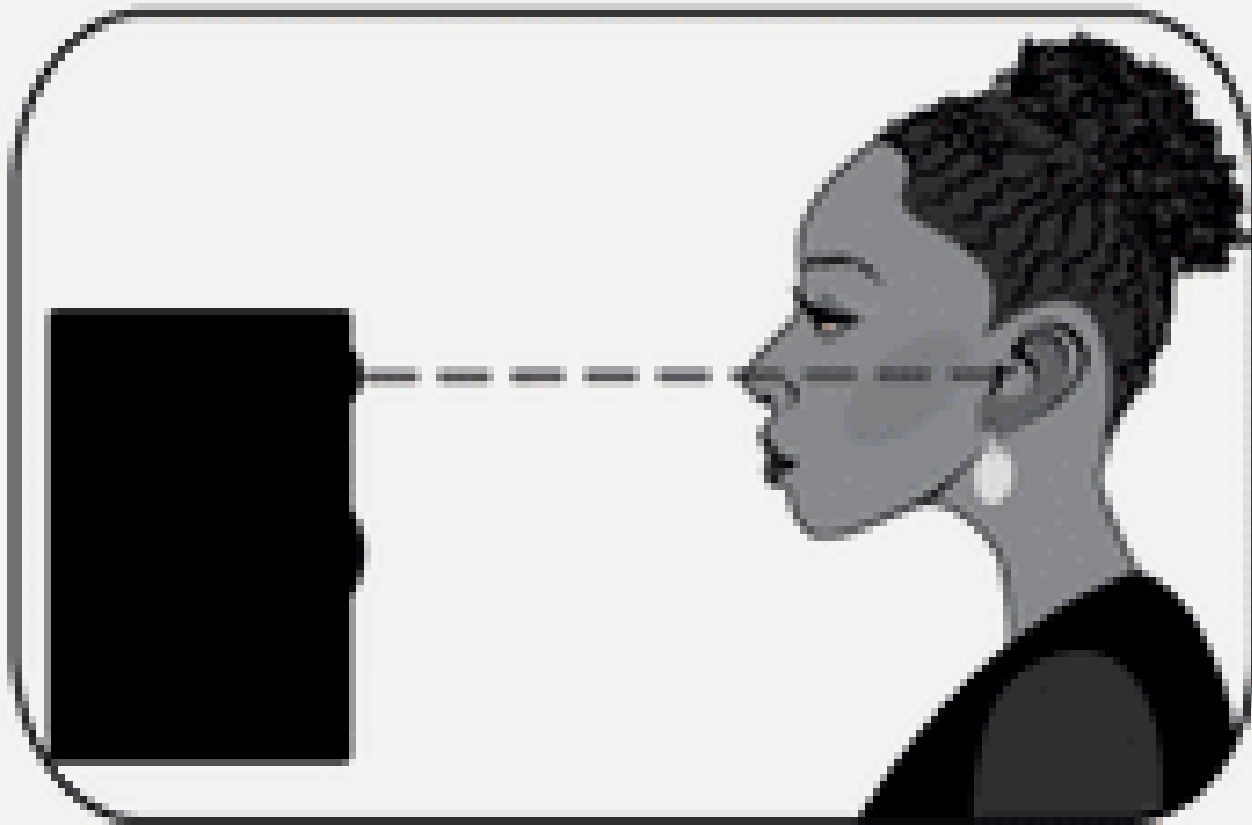
- Placing your speakers directly on your desk can limit their ability to produce clear, balanced audio because the sound waves coming from them are bouncing off a hard, reflective surface (your desktop) before they reach your ears. Studio monitors also transmit their vibrations to any surface they are resting on, including your desk. This can lead to loose screws rattling or other less-obvious noises that can muddy up your mix. Furthermore, your desk will most likely have a resonant frequency or two, so as you turn up your monitors, the desk itself will boost particular frequencies by sympathetically resonating with the vibrations of your monitors.
- Desktop placement also puts most speakers below ear level, which is not ideal. Monitor stands can raise the speakers closer to ear level and help prevent early reflections from interfering

with your listening environment

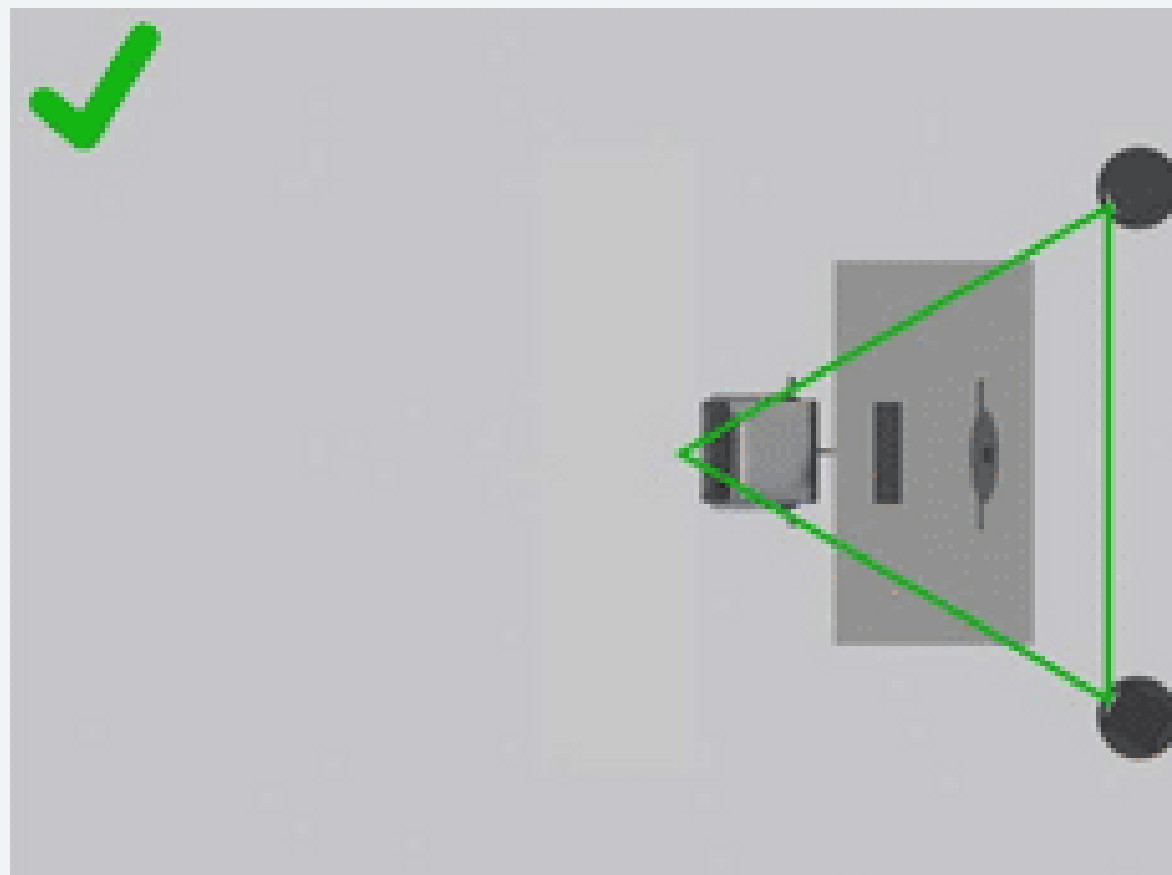
- If you're working in a tight space (or on a tight budget), and don't have the square footage (or spare change) for conventional speaker stands, your speakers may need to be placed on your desk. This is where isolation pads come in. Generally, isolation pads are relatively cost-effective foam or rubber stands for your monitors that help to mitigate the vibrations and sympathetic resonance that can occur whenever a speaker is resting on a hard surface. Monitor pads solve this by decoupling the speakers from the desk. The monitor's vibrations travel harmlessly into a flexible, absorbent material, instead of through and off your desk.



- The high-frequency driver should be the same height as your ears.



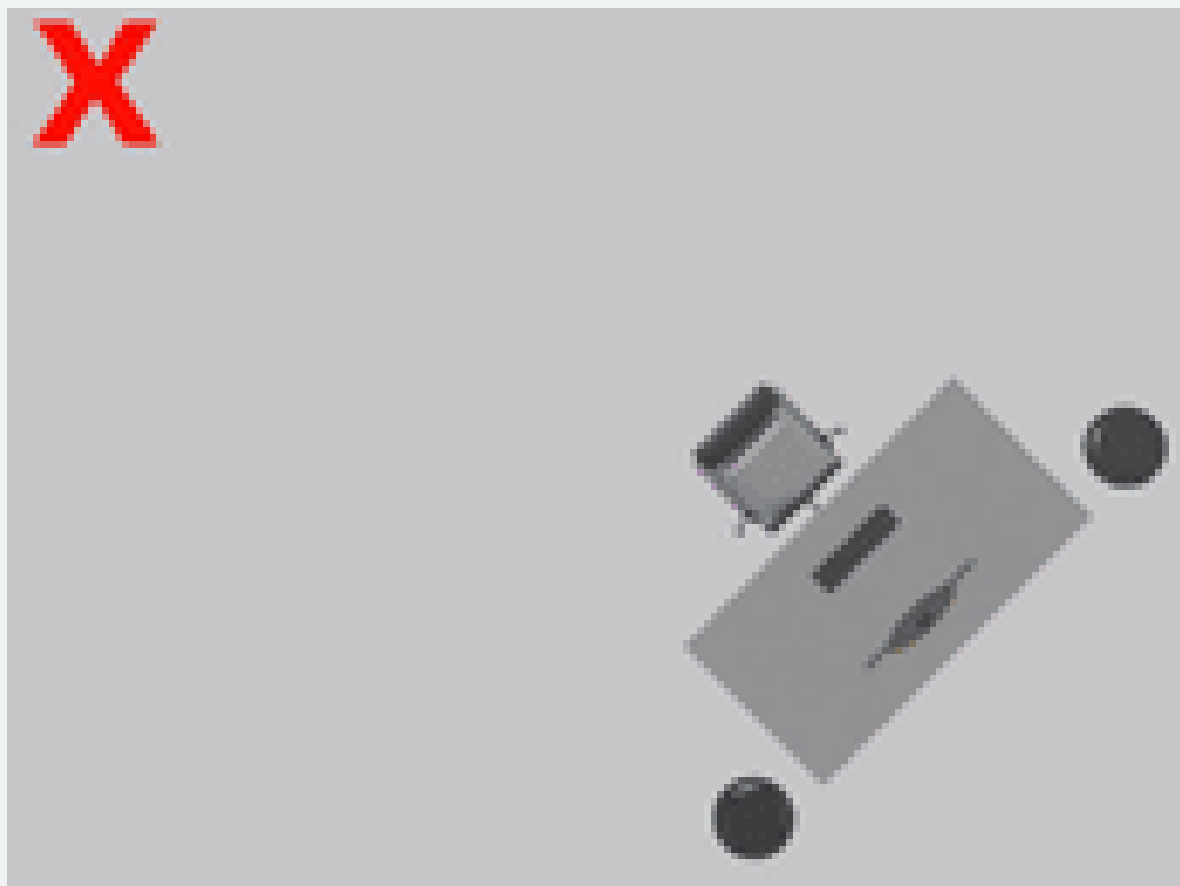
- The Points A,B,C should form a right angled triangle



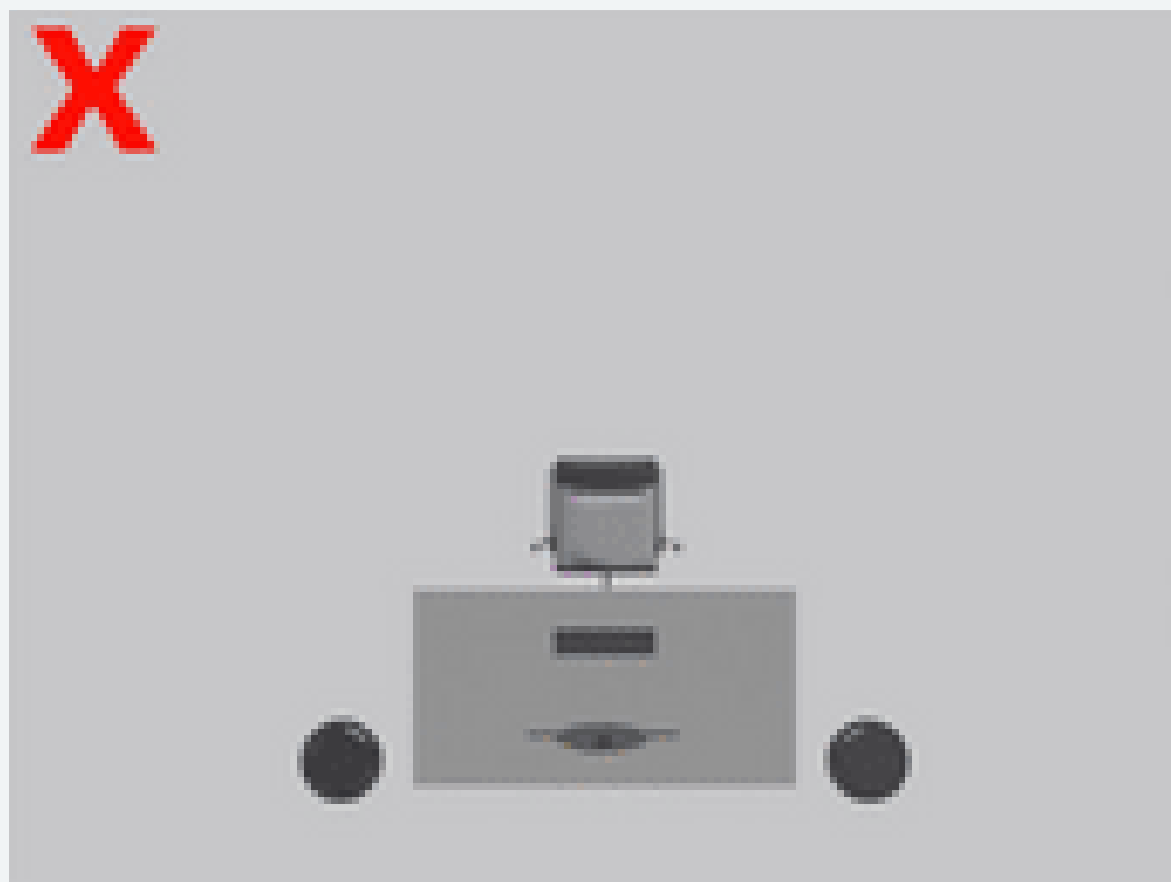
## POSITION OF THE TABLE

Take care of the following while setting up your table position:

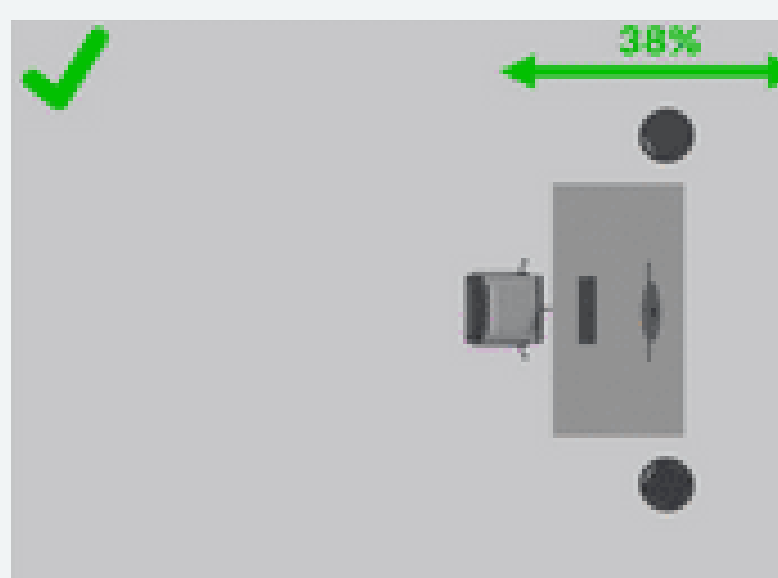
- Your desk should not be at the corner of the studio



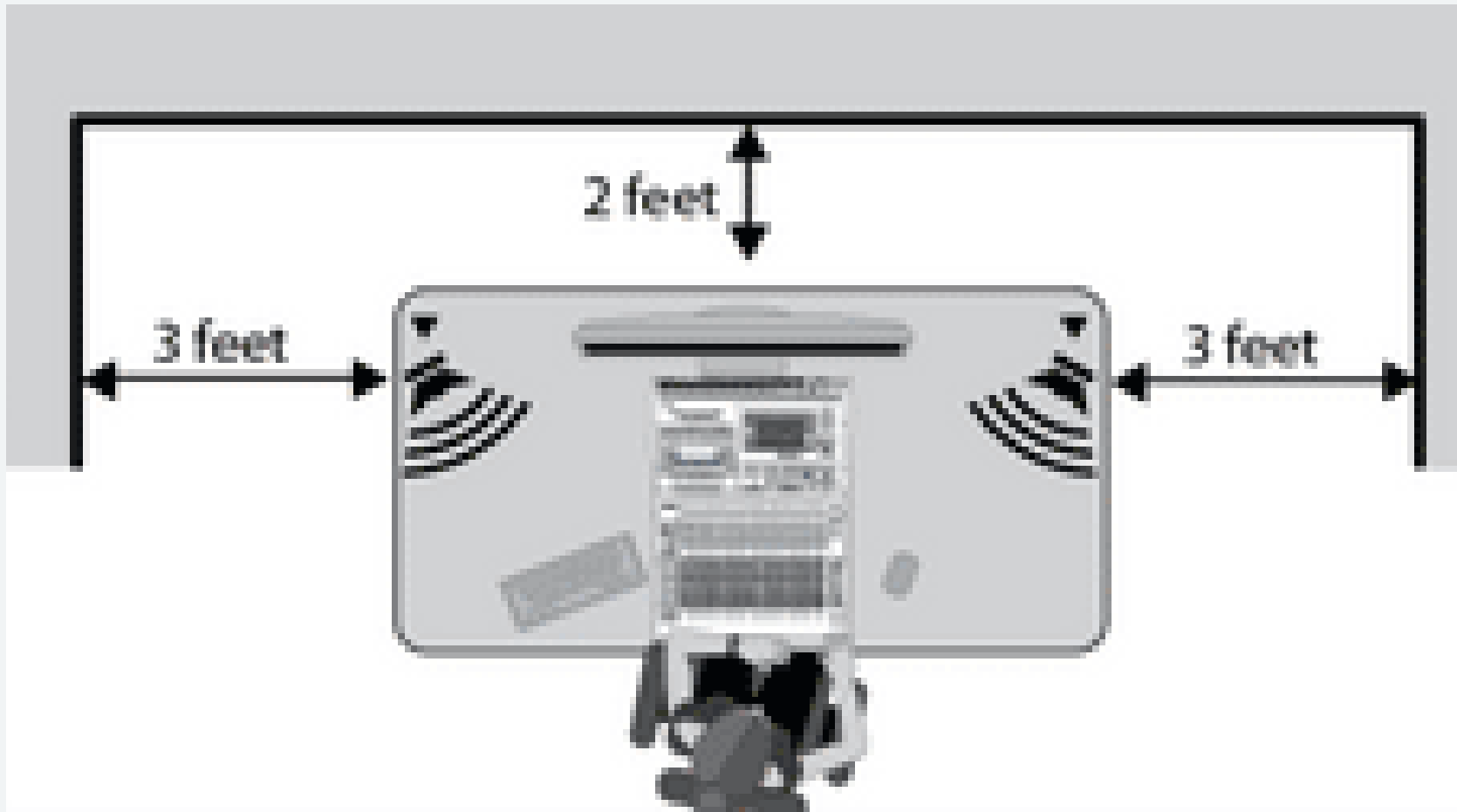
- Your desk should not face the wall of the long side of your studio.



- Your desk should be placed perfectly facing the center of the small wall of your studio and also should not be at the center of your room but closer to the small wall as shown in the image.




- The distance of the desk from the side walls should be more than that of the walls your desk is facing. This image can help you.



These are the most important things you should consider before doing acoustic treatment of your studio. Once you are all set up with setting up your desk and studio monitors you are good to go with applying acoustic solutions to your studio.





# ACOUSTIC TREATMENT OF YOUR STUDIO

04

Here we will look up on how to set up acoustics in your home studio for a better environment of mixing.

Acoustic treatment is one of the most essential parts of your music studio; it allows you to trust your ears. So I'll be showing you how to treat your home studio effectively.

It's true that untreated rooms aren't ideal for mixing. For example, common acoustic problems can lead to perceiving multiple 12+ dB resonance boosts in the low-end of your mix, hearing echoes, and perceiving notched filtering effects throughout the frequency spectrum of your song. Mixing in an untreated studio is like driving with a blindfold on.

When your speakers produce sound, there are three distinct stages in which you perceive it. First, you hear the direct sound coming straight from your speakers, then you hear the early reflections that have bounced off boundaries like your desk, walls, floor, and ceiling. Finally, you hear the sound's reverberant field, which is a complex network of reflections created via the original sound interacting heavily with your room.

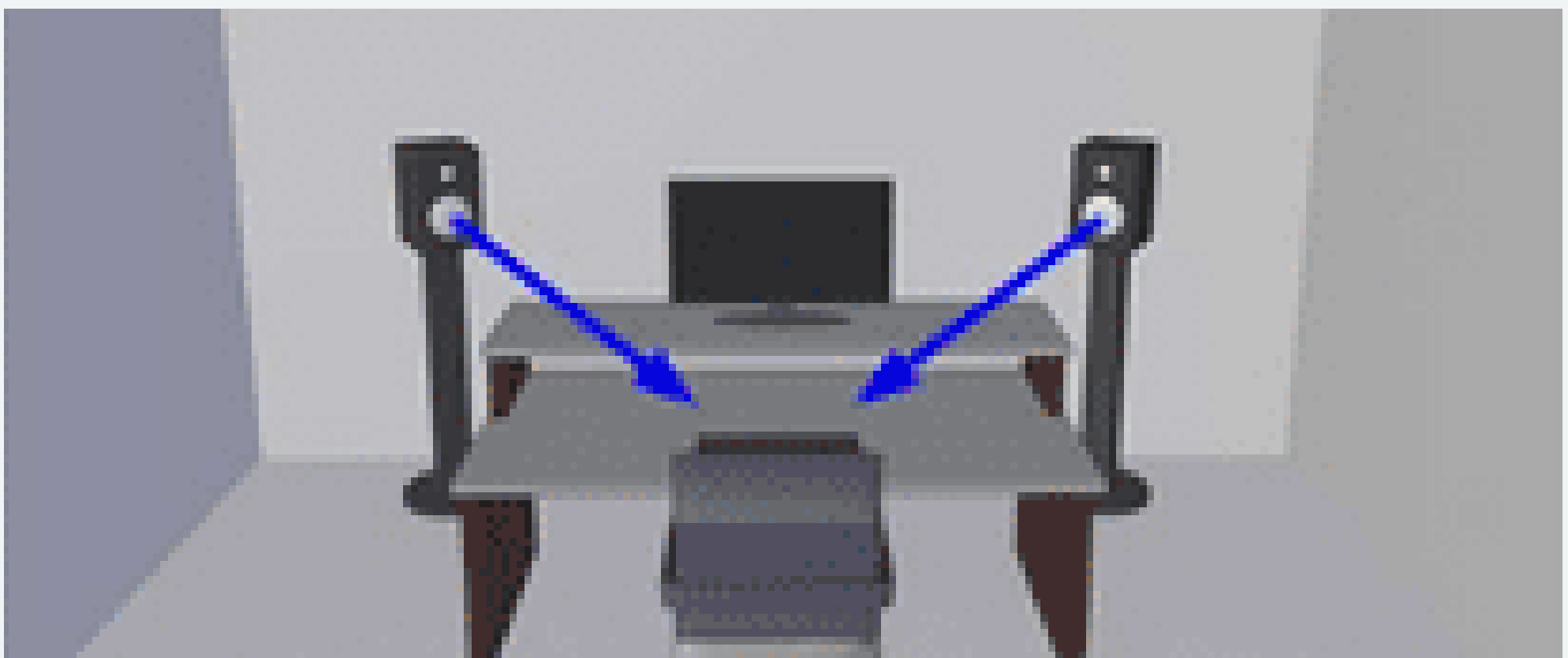
The four most problematic areas in your studio include the ceiling above your desk and the walls to the left and right of it, the corners of your room, the parallel walls to the rear of your room, and the back wall of your studio.

Applying appropriate acoustic treatment to these critical positions can significantly reduce acoustic problems.

But first, let's understand in how many ways our ears perceive sound coming from speakers:

# HOW WE PERCEIVE SOUND

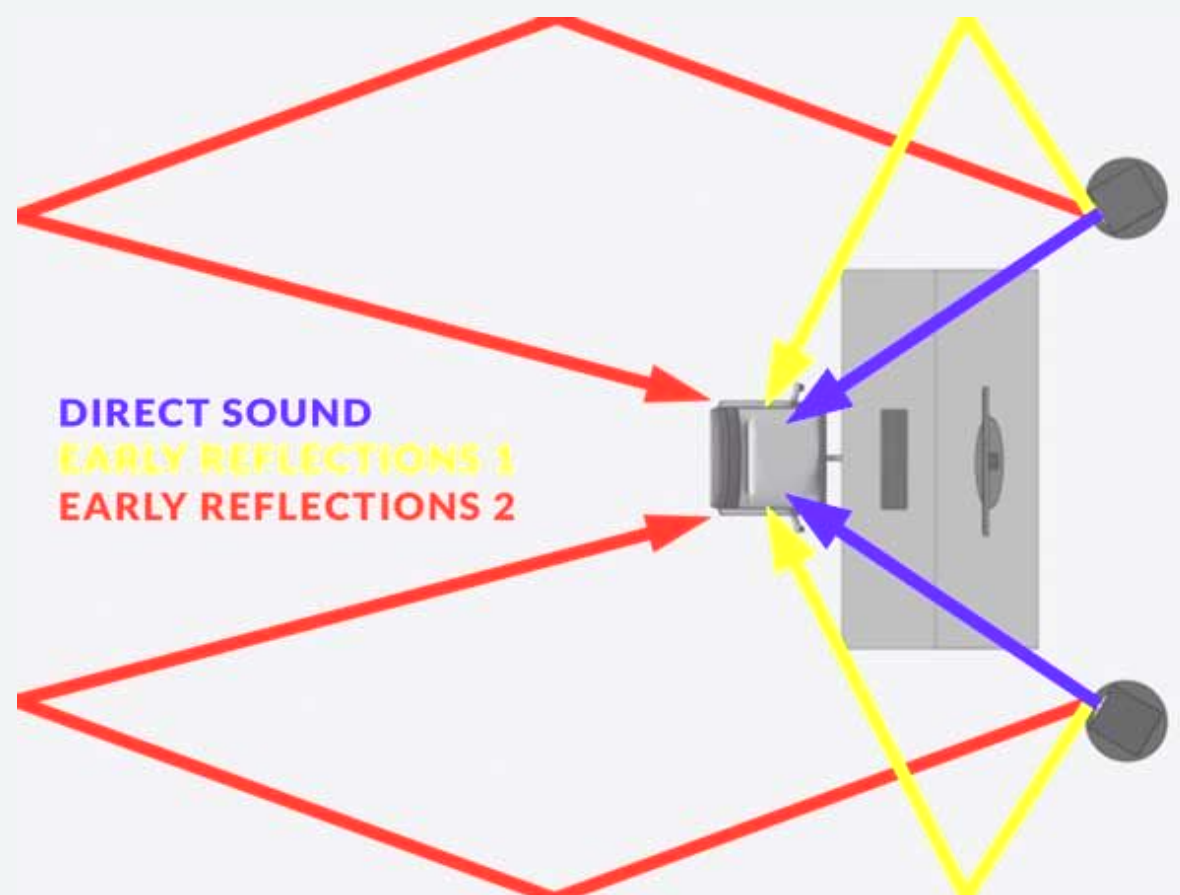
## 1. DIRECT SOUND



The direct sound from your speakers will arrive at your ears before any of the room's reflections. This initial wavefront is the most unhindered version of

your mix. It's in your best interest to create a clear separation between the direct sound generated by your speakers, and the early reflections created by your room.

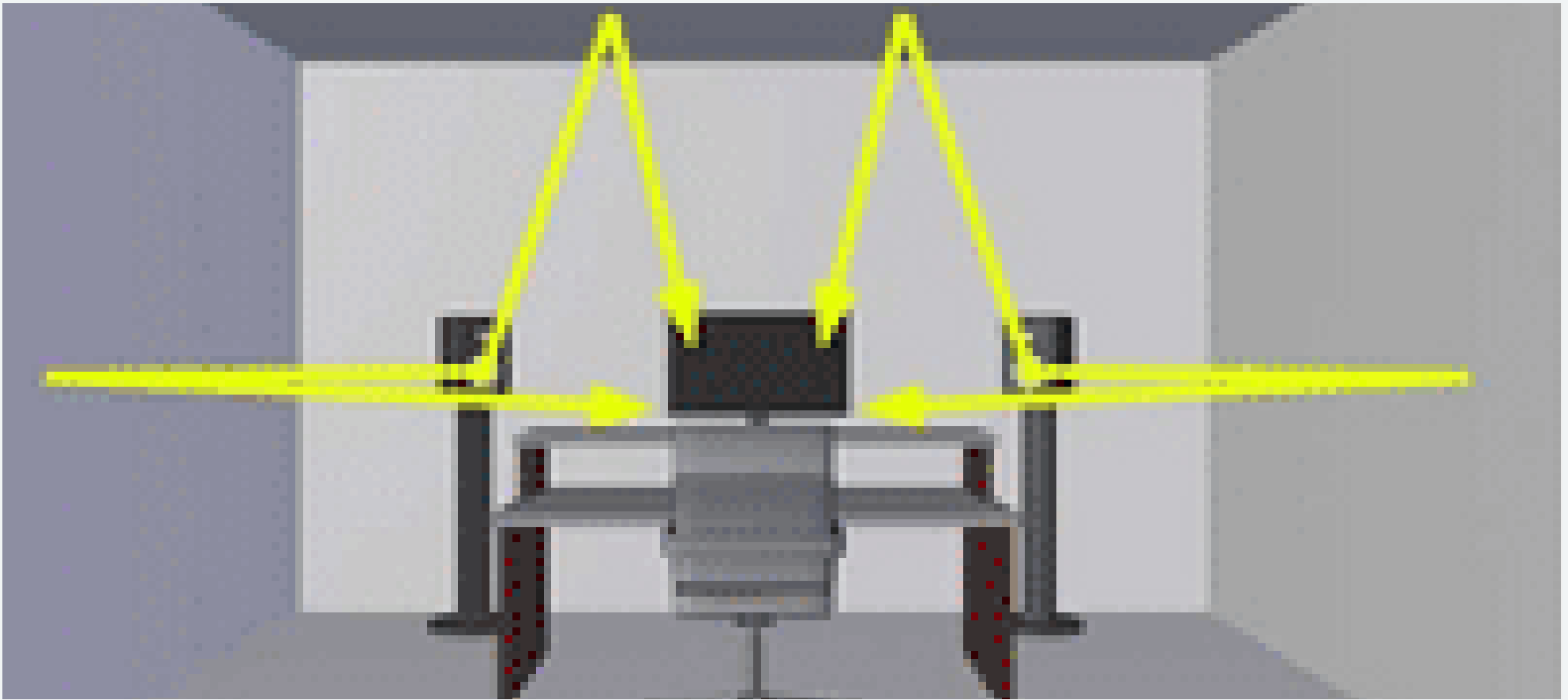
## 2. EARLY REFLECTIONS



Early reflections arrive at your ears soon after the direct sound, with the first early reflections bouncing off the walls to the side of your desk, as well as the ceiling above it.

Early reflections can end up reflecting off the back wall of your studio too.

Depending on the orientation of your desk and speakers, you can also experience early reflections bouncing off the flat surface of your desk.



This can be a really big problem when you have a pretty larger room/studio and you want to work with studio monitors

So up next we will see how to get rid of these unwanted reflections by doing the acoustic treatment.

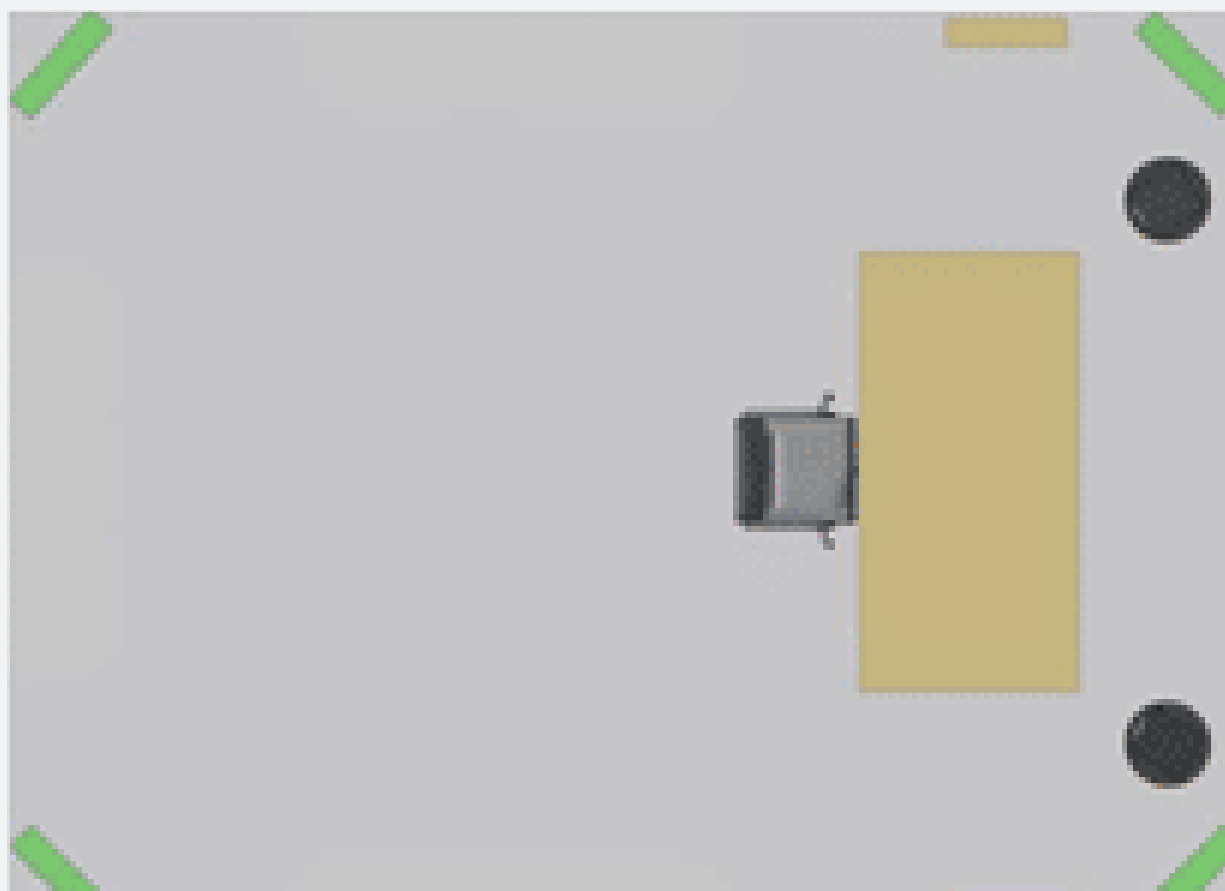
# APPLYING ACOUSTIC TREATMENTS

## 1. CORNERS

Room modes end up in corners, so this is where bass builds up. Placing bass traps in the corners of your room helps to convert this low-end energy into heat.

Bass traps come in either the form of panels or wedges that butt up into the corners of your room. A common place to install bass traps is where your walls meet one another, but some people also choose to apply bass traps where the walls meet the ceiling, and/or where the walls meet the floor.

The acoustic treatment in the following image is what I consider to be the minimum number of bass traps that you should apply to your studio. Foam wedges from Amazon don't work well as bass traps because they aren't dense enough to absorb low frequencies. For a better solution, check out Tri-Trap Bass Traps.



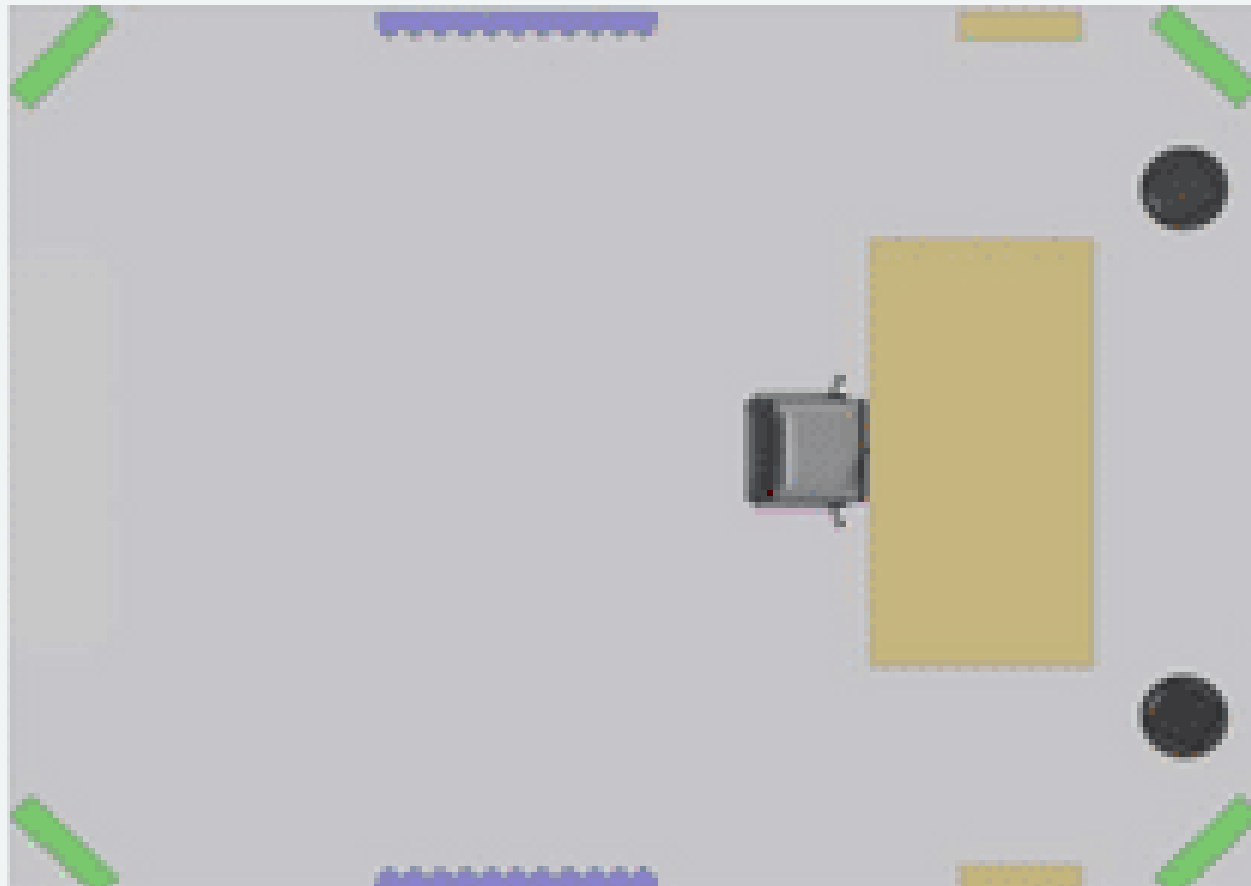
## 2. REAR SIDEWALLS

Exposed parallel walls, like the rear sidewalls in your studio, are something that you want to avoid because they can cause flutter echo. Installing reflectors/diffusors like the Q7d Diffusor in this critical zone will scatter the sound that hits them, and prevent flutter echo from occurring.

If you find that your studio still sounds too lively with reflectors/diffusors in this position, you can try installing absorbers instead.



However, if the absorbers don't deal with the flutter echo, you may want to take a look at hybrid diffusor/absorber options.

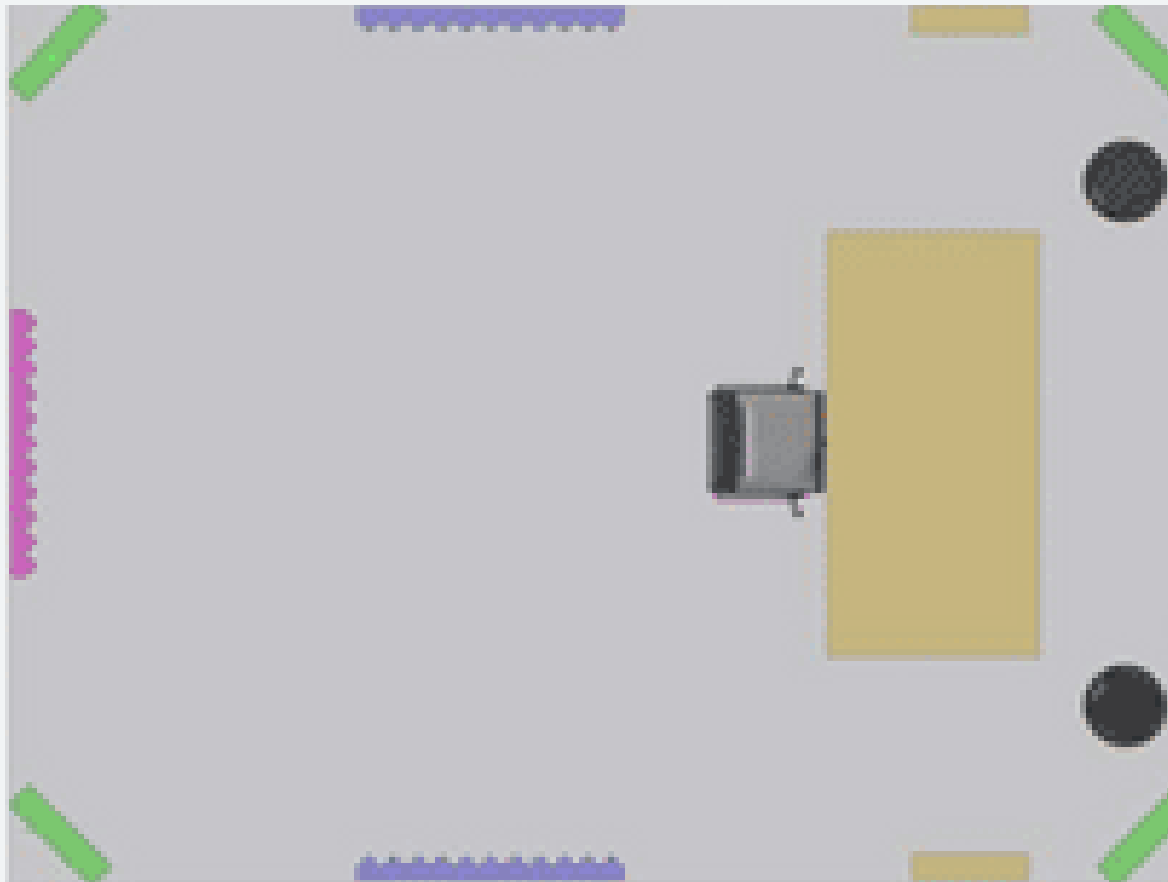


### 3. BACK WALL

The back wall and the front wall in your studio are also parallel with one another, so flutter echo is a potential issue yet again. On top of this, standing waves are of particular concern. Having moved your desk around, you should have already minimized the effect of standing waves to the best of your ability, but there is the acoustic treatment you can apply to the back wall of your studio to further reduce the adverse effects of standing waves.

Your back wall is where you'll definitely want to make use of thick hybrid acoustic treatments like Monster Bass Traps. Absorption will reduce the energy of bass frequencies significantly lessening

adverse effects of standing waves and reflection will help with diffusion, effectively dealing with flutter echo.



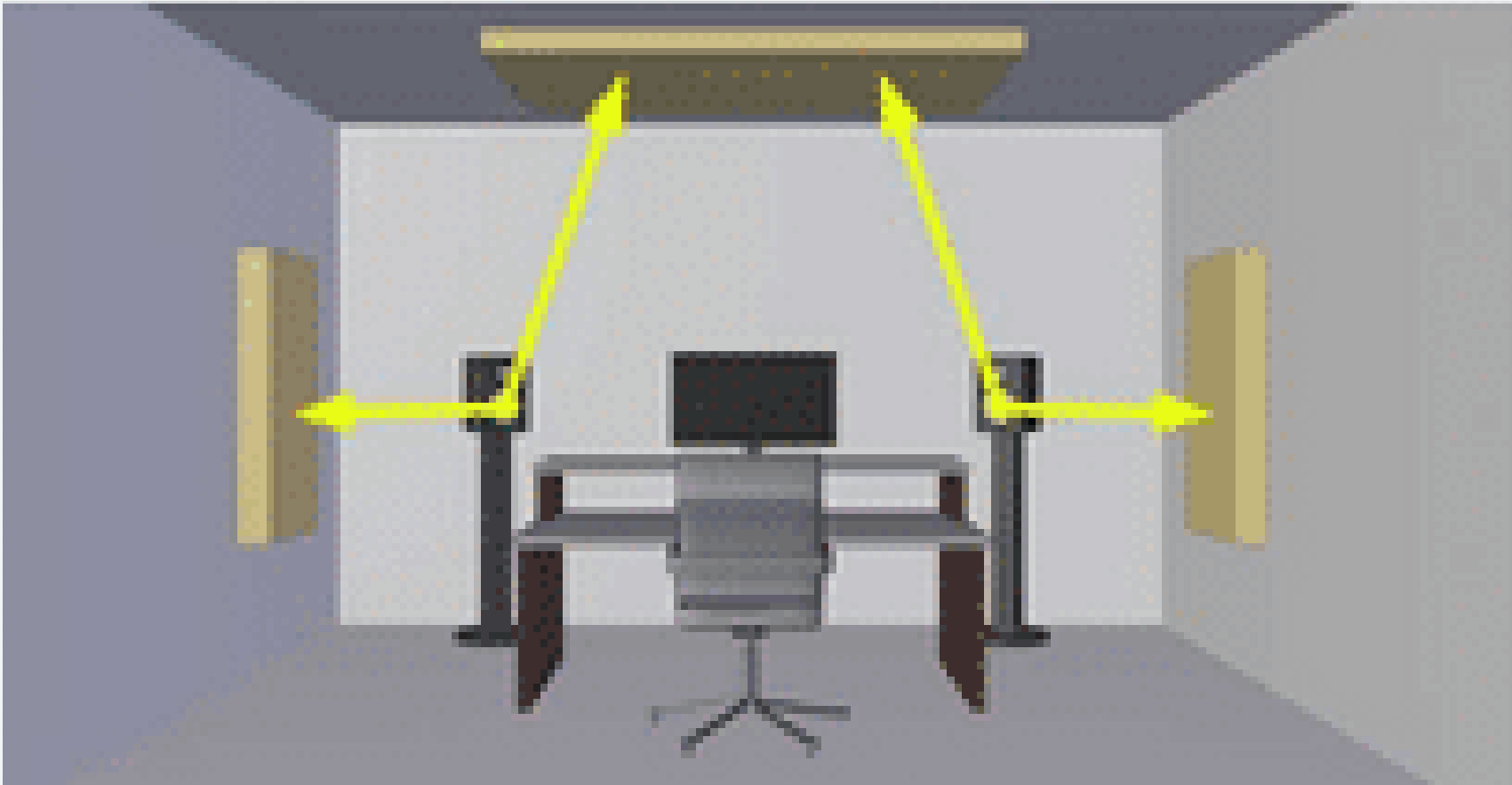
Some people choose to apply acoustic treatment directly behind their computer monitor, which is fine, but I would argue that treating your back wall is of a higher priority. By the time sound travels to your back wall, and reflects back to the front of your studio, the damage has already been done; untreated low-end frequencies will have already created standing waves, and your perception of the bass in the music will already be skewed.

## 4. CEILING

You want to apply acoustic treatment to your room in a way that creates a clear distinction between the direct sound and early reflections. This will help to reduce comb filtering. A popular method of dealing



with early reflections is through the use of absorbers.



To deal with early reflections from your ceiling, you can suspend a large absorption panel halfway between your listening position and the speakers. Mounting bracket kits that are used to suspend acoustic panels are pretty cheap, and installing these ceiling panels, or ceiling clouds, is a rather simple undertaking.

So now we are almost done with the things that you should do before mixing. Now we have our studio monitors perfectly placed and our studio perfectly treated. So now it's time to learn about actual mixing.



# PREPARING FOR MIX

05

The first thing we need to do before touching any knob is to make sure that our project is ready to be mixed i.e we have to prepare the session for mixing. This is the first step towards mixing and by the end of this section, we might actually have our project sounding a little bit better by just preparing it correctly.

We're gonna get rid of some noisy stuff, tune some things, and we're going to set up a couple of effect buses and run some stuff through some compressors and limiters.

So just by doing this process of setting up the session to be mixed We probably will end up with a better sounding track before we even get into the nitty-gritty.

## **1. COLORING AND NAMING TRACKS**

It may look tedious to color and name every track, but trust me it will help you to navigate easily and will speed up your mixing process. Suppose while mixing you felt like the kicks you used isn't that punchy and since you have colored your drums and red you will be able to find it in seconds. We have created a color code for audio tracks you can use.

DRUMS	RED
PERCUSSION	DARK RED
BASS	BLUE
KEYBOARDS & SYNTHS	GREEN
GUITARS	DARK GREEN
LEAD VOCALS	PURPLE
BACKING VOCALS	YELLOW
STRINGS	CYAN
HORNS	BROWN
AUX, BUS, MASTER	ORANGE

Of course, you can use color according to your taste but these are kind of standard and used by most people, it can help you in collaboration and presentation.

## 2. TAKING BACKUP

Before tweaking any knobs the first thing you should do is to take a backup of your unmixed project. Because when you start mixing you may test various effects and it sometimes sounds good but sometimes it really destroys the whole feel and after lots of tweaking you will find yourself in a situation where you want to roll back to the previous version, this is when a backup will help you.

So make sure you take backup every day when you do some editing.

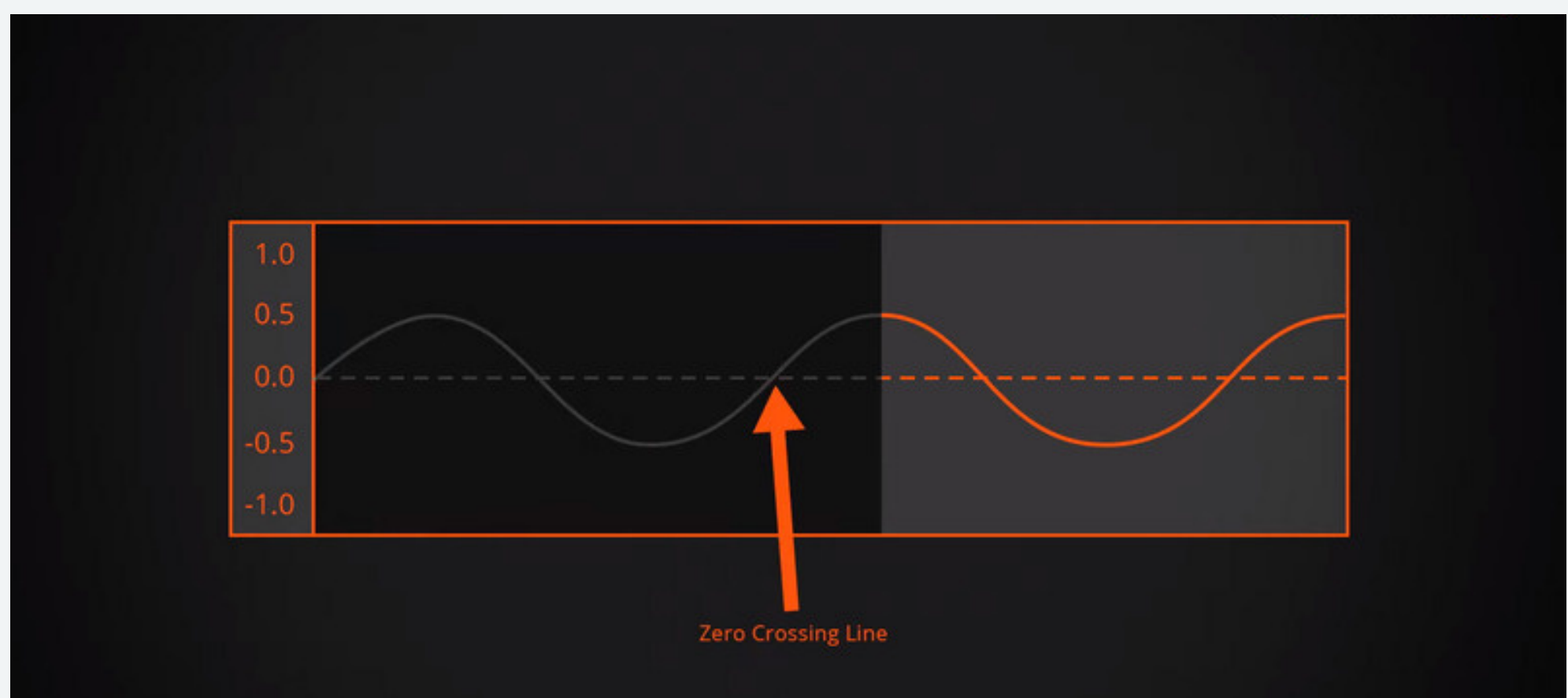
### 3. EDITING AUDIO SAMPLES

There are a few tasks you should do before using any plugins on your audio plugins to get more out of your effect plugins by cleaning the tracks.

While adding some samples like guitar, vocal shouts, drums, etc. you may sometimes hear clicks or pop sounds while playing it with the rest of the tracks, this is because of the broken waveform.

There are two ways to fix this

#### A . CUTTING AUDIO SAMPLES AT "0"



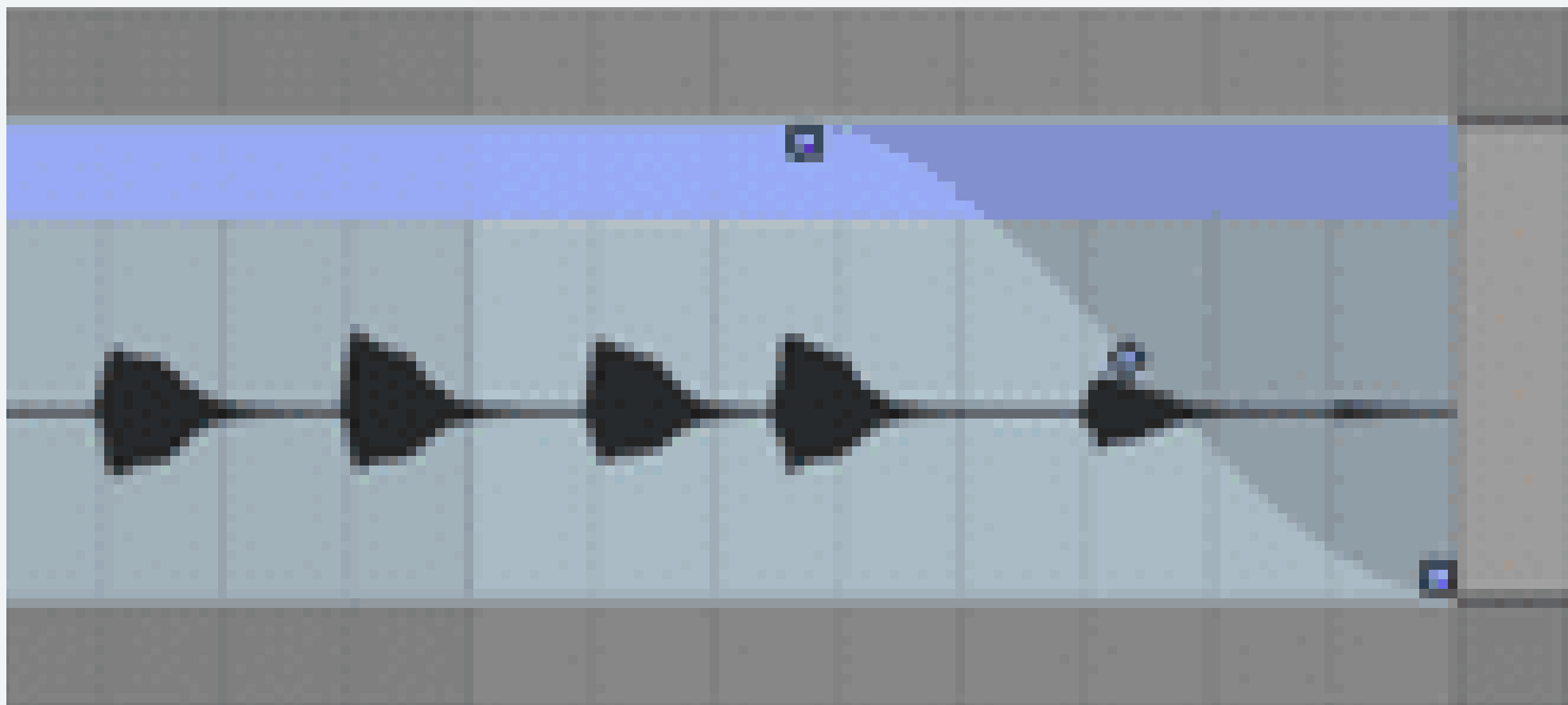
When cutting audio, you should try and aim for a "zero crossing" point aka where the audio crosses that center line.

This is easy to find/do when a signal is mono

because there's only one waveform to cut (even if it's a mono signal split to an L&R channel they are mirrored).

Cutting Audio becomes a little trickier with stereo information because the Left and Right channels might not be crossing the "Zero" at the same point...

Using FADES is ideal in these situations because it allows you to smoothly fade in, or out, of your audio. It will literally make your audio start or finish at a zero-crossing point.



## **B . REMOVING EMPTY SPACES**

One of the best things you can do to your tracks to look more focused is to cut the empty space between vocal phrases on any loop which has spaces with literally no sound.



You should also remove breathing noise from vocals if you don't feel like making it a vibe.

## CREATING EFFECT BUSES FOR MIXING

What is a Mix Bus in Simple Terms?

A mix bus is a channel through which you send your instruments and audio channels. So when you apply an effect to the bus track, it adds that effect to every channel routed to it.

The best metaphor for a mix bus is an actual bus.

Imagine that all of the channels in your DAW are people. And the mix bus channel is an actual bus.

You can tell certain people (channels) to get on the bus. Then any direction the bus goes, the passengers (channels) will go.

For example, if you send the bus down the reverb road, then reverb will be applied to every channel on the bus.

Technically, a mix bus copies the signal of a channel,

applies the effect, then adds the signal back into the mix.

## **MIX BUS VS MASTER BUS (MASTER TRACK)**

Just to be perfectly clear, a mix bus is not the master bus (AKA the master track).

If a bus track is a bus full of children, then the master bus is a ferry that carries those busses and school children.

Every channel in your DAW is fed through the master bus.

As a general rule, you don't ever want to add effects to your master bus. This will screw up the track when you bounce it for mastering. Just don't touch the master bus.

Now you may question why go to all the trouble of creating a mix bus track?

Well, because a bus track copies the signal, it leaves the original signal untouched. So you have more control over both the original and the duplicated one. Let's look at this practically...



If you want to add reverb to your vocal. You could slap the reverb effect directly to the vocal track and that would be fine.

But there's a problem with this.

If you want to change the amount of reverb (i.e. the wet vs. dry signal), it's difficult. You have to open the plugin and edit it.

This is a real problem when you have multiple vocal tracks with the same reverb effect. You'll want to change the reverb settings for all of them or for each of them.

Instead of opening the effect on each individual track, you can change the setting on the bus track, which will change the effect on all of the tracks. Or you can quickly adjust the amount of reverb on each track from the bus track.

Using bus tracks will make your life way easier and your mixes so much better.

## **HOW TO SET UP A MIX BUS**

The most common effects to use with a mix bus are

compression, reverb, delay, and EQ. For each of these, here are the steps to create a mix bus:

- Create a new track
- Title it “[effect] bus”
- Route the desired channels to this bus track (choose “send” from each channel or add a “receive” of each channel from the bus track)
- In Ableton, it’s pretty easy to make an effect bus or (return track)

Now here are some tips for bussing those four effects.

## 1. MIX BUS COMPRESSOR

Mix bus compression helps your tracks feel more together. It keeps the levels consistent the whole way through and it makes things sound more exciting.

Just as everything in mixing, mix bus compression works best when it’s subtle. Try these settings:

- Use a ratio of around 2:1 to lightly compress the tracks
- Start with a slow attack (try 30 ms) and a fast release (keep it under 50 ms)

- Don't apply any more than 2 dB of compression
- This works best when you apply the bus compressor before you start mixing. If you mix your track and then add a bus compressor, it could completely change the dynamics you worked so hard to get.

## 2. REVERB & DELAY BUS

If you want to add a certain reverb or delay to multiple tracks (like your vocals), you can use a bus reverb and/or delay. These effects should be on separate bus tracks. You'll have more control, and it's easier to adjust the settings for multiple tracks all at once.

But there's another method that can add space and width to your mix.

An engineer friend of mine taught me a bussing trick that has totally improved my mixes.

Here's what you do:

- Create two new tracks: a "Reverb Bus" and a "Delay Bus"
- Apply your favorite reverb effect to the reverb

bus and your favorite delay effect to the delay bus

- Route every single track in your DAW except for the bass, drums, and percussion to each of these bus tracks
- Pull down the gain levels of the bus tracks to the point where you almost can't hear the effects working
- This can add depth and spaciousness to your entire track. But remember, be subtle with it.

### **3. EQ BUS**

To be clear, you need to do the main EQing on each individual track. Each instrument can have different frequencies you need to cut or boost.

Mix bus EQing is helpful for making small tweaks to a group of tracks as a whole.

If a group of vocal tracks together create an overpowering feel, you can bus them to an EQ track and cut or boost. Or if your cymbals need a bit of shimmer, use a mix bus EQ.

Like every other part of mixing, stick with small movements, especially on a mix bus. One little change

in the frequency spectrum can have a big effect.

So use wider bands and smaller cuts (no more than 2 or 3 dB). This is to shape the tone, not to do surgery. Do surgery on the individual tracks if needed.

## **COMMON MISTAKES WHEN USING A MIX BUS**

Mixing is a subtle art, and that's especially true when you are using bus tracks. So it's easy to make mistakes. That's why I want to cover the most common mistakes people make when using a mix bus.

### **Mistake #1: Wasting Money On Plugins**

You don't need to spend any money on plugins to use a mix bus. You can use stock plugins and mix a track to industry standards.

I'm not saying you should never upgrade, but the more important focus is learning the technique of a mix bus. And for now, you can use a stock plugin.

### **Mistake #2: Making Big Adjustments**

I'll say it again: mixing is all about little adjustments. And when you're making adjustments to many tracks

at once, you'll hear the change even more.

Here are some tips for making small movements:

- Compression: use a 2:1 ratio
- Compression: shoot for 1-2 dB of gain reduction
- Compression: use a slow attack time (start between 30-50 ms)
- Compression: use a fast-but-not-too-fast release time (keep it under 50 ms)
- EQ: use wide bands
- EQ: make small cuts and boosts (no more than 2-3 dB)
- Reverb & Delay: keep the levels lower than you think they should be so they don't overpower the mix.

### **Mistake #3: Adding A Compressor At The End**

I briefly mentioned this, but using a mix bus compressor only works if you're mixing with the compressor already applied.

Compression can change the dynamics and tone of a track, so adding it after you've already mixed the song can ruin your mix.

Before you start mixing, add a mix bus compressor.





# THE ART OF MIXING

006

In this section, we're going to talk about the main elements of a good mix.

Now, this is really important to understand, because, for the rest of the course, we're gonna work through how we build each of these elements into our mix.

There are six elements that you want to have represented in a mix

### **1. The 3 Dimensions of Sound**

### **2. The 3 Dimensions of Music**

There's a lot of different ways to approach mixing but this is how I learned to mix, and this is the way that a lot of people do it.

Consider mixing as an art. There's nothing like a formula that I can give you like turn this down, or turn this up, and at the end, you'll have a great sounding mix, but there are certain things that we will do in order to make things sound better that are usually true in most cases. But in order to make a great sounding mix, you're gonna have to work in the same way that a painter works.

You have to imagine your project like a painting,

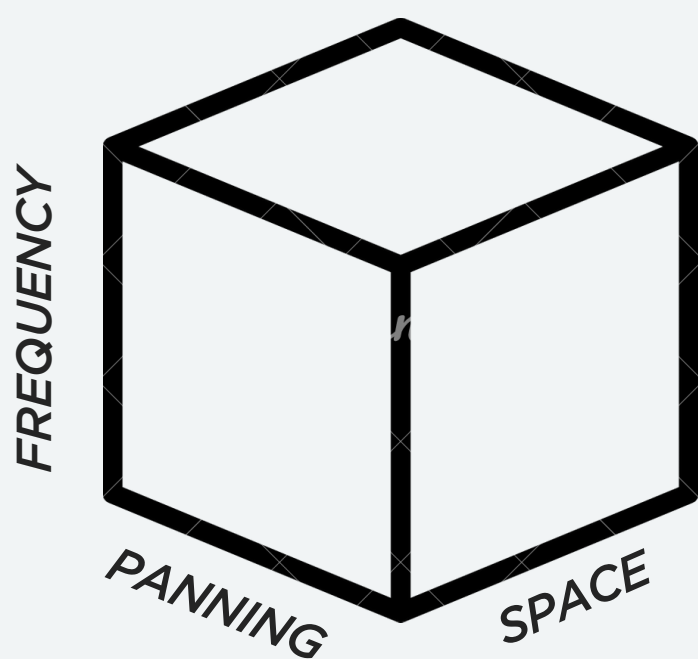


and understand that there is either a good song or a bad song, there's nothing like a correct song, so don't get overwhelmed with something that doesn't exist like a perfect mix.

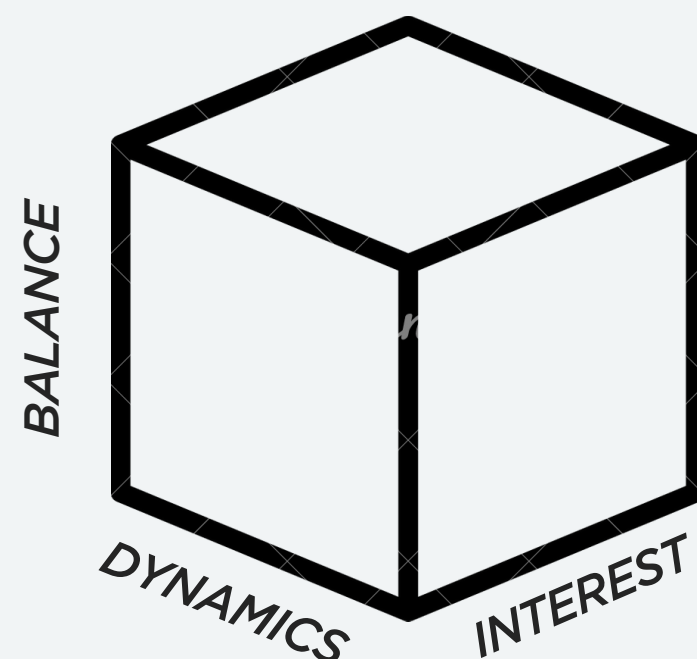
While mixing you should use your artistic mind, because all you have to focus on is how to make a GOOD sounding mix and not a CORRECT mix, and when you'll understand this you will feel more confident while mixing your songs.

So there are three dimensions of sound and three dimensions of music, so let's just talk about the differences between sound and music.

### 3 DIMENSIONS OF SOUND



### 3 DIMENSIONS OF MUSIC



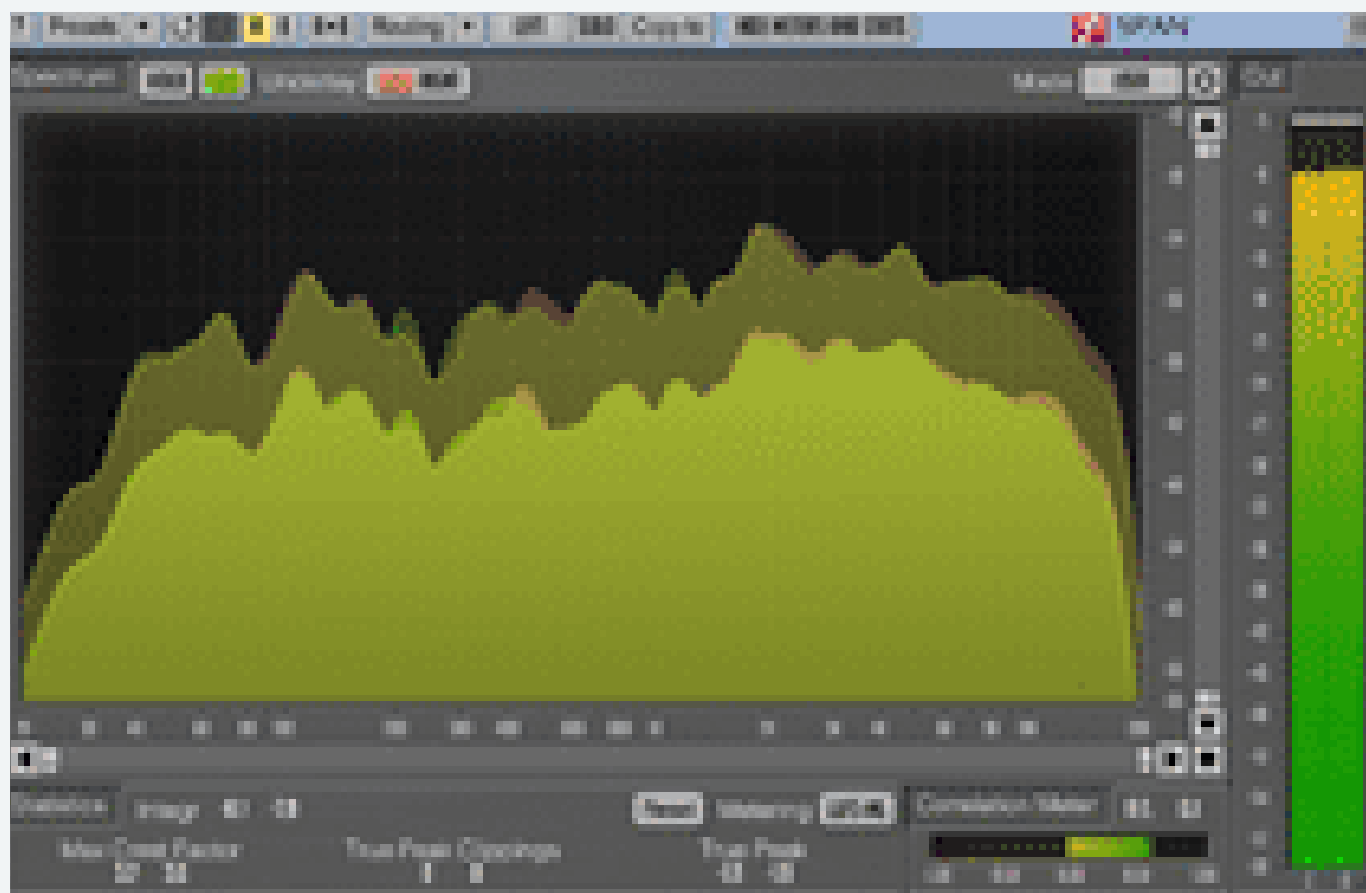
## THE 3 DIMENSIONS OF SOUND

So there are 3 dimensions of music, frequency, panning, and space, Let's learn about them one by one:

# 1. FREQUENCY

While mixing, some people think their job is just to make one thing louder than the other. Right? but that's balance, and that is different. That's a musical element and we'll learn about it further. What we are focusing on here is frequency. What we need to do here is basically make sure that all frequencies are easily represented.

Let me show you what that might look like If I just take a spectrum on a master channel. There are a lot of different spectrum utilities you can use, like SPAN.



This is how it will look when you put that in any of your tracks. So in the vertical here is the volume and in the horizontal is the frequency.

Here you can see that the low and high ends are missing.

This is what we have to do. We have to craft this a little bit, this shouldn't curve all the way down like this. So we'll look at how this works later.

That's one element of what we're going to be working and we will make sure that all the frequencies are coming through, and there's a good distribution all the way across of all the different frequencies.

## **2. PANNING**

Panning is moving from left to right channel. If something moves between the left ear or the right ear, we call it panning, that's something you can do to make your sound wide.

Let's go back to painting a picture analogy. You should make a picture that takes up the full canvas right? Not just one little spot on the canvas, and we do that with panning.

## **3. SPACE**

When you use reverb on some vocals and you turn the mix (dry/wet knob) to almost full, you will see that the vocals seem to be far away from your ears, and more surrounded. This is how you can use

reverbs and other effects like reverb, stereo widener to add a lot of space in your mix.

## THE 3 DIMENSIONS OF MUSIC

Now there are also three dimensions of music and the ones that we are concerned with when we are mixing are balance, dynamics, and interest.

### 1. BALANCE

Balance is kind of loudness of every track in our project. The most important element in any mix is the balance between each track. By using simple level manipulation, you can quickly dial in a balanced mix. Doing this first leads to less significant changes when processing your mix with plugins, meaning you can get greater results with even the simplest tools.

What needs to be loud, what needs to be quiet is an important thing to consider, making sure that the balance is right, making sure that we hear everything that we want other people to notice.

While balancing you should ask yourself if you want to place the vocals in front of the guitars or if you want to hear the solo guitar in front of the backing

guitars, etc. Just aim for what you want to achieve before starting to balance as it can also take your most of the time.

## 2. DYNAMICS

When we alter the volume of a particular sample at different places, we are playing with its dynamic, which means we are changing its dynamics. We have to look if the chorus is louder than the verse? Should the bridge get quiet and be quieter than the chorus? these are a few dynamic questions that we're gonna ask.

So when we ask, are the vocals louder than the guitar? We are questioning about the balance. The dynamic question is, for the particular track or sound, should the chorus be louder than the verse for the intro? Or it should be quieter than the first verse? Things like that bring dynamic contrast to your mix and that's important.

If the sounds in your project are not balanced, whether it's dynamically or comparatively, then you're just gonna exhaust the person listening to it. They will say "Oh my gosh, I can't hear anything clearly." So dynamics are an important thing. Now the third element is probably the trickiest one and it's...

### 3. INTEREST

#### So what is INTEREST?

Interest is finding something unique in that music, finding the thing that needs to be the focus, finding that interesting point, and then making sure that it is clear and comes to the front.

The thing that is the most prominent in the mix, that's the element of interest, we'll work on that later in this guide.

So those are the three dimensions of music, combined with the three dimensions of sound, we have six main things that we need to work through.





# MIXING

## ESSENTIALS

(GETTING STARTED)

# 07



Before starting, I want you to understand what things you should keep in mind and make you understand what a good approach looks like.

## **THINK SUBTRACTIVE**

Now, this is the thing that creates a difference. The first thing I want to point out is the mindset you should be in. While mixing a piece of music, your mindset should be subtractive. What that means is you should think about taking away rather than adding things to your music.

Most of the time, when people make tracks, including myself, they add a lot of elements to beef things up. And while mixing, you might find that you don't need all of those elements. It just needs to be mixed a little bit better. So sometimes you should pull out some elements and then try to make the elements that are already there really prominent in the mix.

The same goes when it comes to volume, we should be thinking about pulling things down rather than pushing things up most of the time. I'll explain that once we get into the setting levels section, but remember subtractive thinking is the name of the

game here, so remove things rather than adding things, pull things down rather than pushing them up.

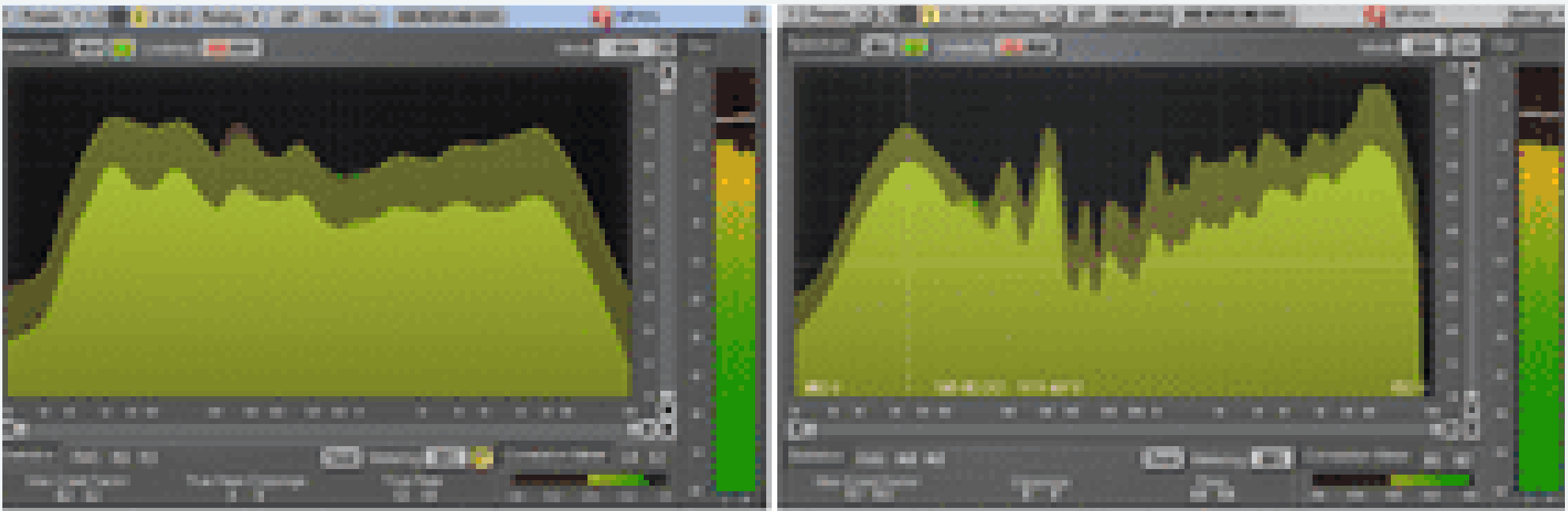
## DEALING WITH FIGHTING INSTRUMENTS

One of the biggest things we have to keep an eye on is conflicting instruments.

We have to listen through our track and find things that are in the same frequency range and have the same volume. Then we have to create a separation between them so that they don't get in each other's way and we have to do this in such a way that we can hear both tracks correctly. We have to preserve the characteristic sound from each track while creating separation.

Now, there are other reasons also that we should keep things separate from each other. We don't want two or more tracks to be in the exact same frequency space, as this can create a muddy sound. Don't worry, we'll deal with that later when we reach the frequency section.

**Let's look at these examples:**

**A****B**

Both A and B instruments have a good presence at the low and high end, so they may probably get in each other's way, So we can separate them by EQing them which we will discuss in the Equalization Chapter.

## **FINDING A STARTING POINT**

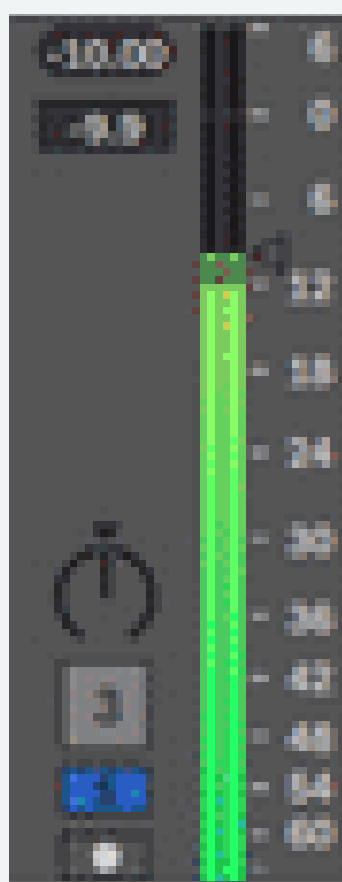
The next we need to decide is from where we are going to start mixing. We can use two methods for this.

### **1. USING REFERENCE OF ANY TRACK**

Now, what I mean by that is, just listen to one element of the track, make sure that the level is set where we want it to be, and then balance everything according to that, and that can be a really good starting point.

If you are working on an EDM track, you may most probably start with a synth or whatever is the element of focus you have in your track or maybe a kick. Another track you can start with is vocals. Get the vocals perfectly balanced right where you want it and then build everything up around it. Sometimes even guitar can be a good starting point, but according to me, vocals are probably the best place to start. Because remember, everything is relative here, if we're going to mix everything to the volume of this kick, it's really important that I put this kick in a good spot.

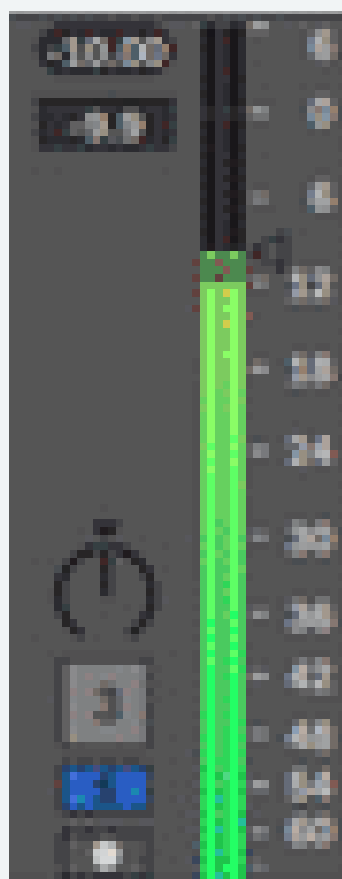
If I choose to start with the kick, I'll lower the kick level to -10db.



Now, let's talk about how to read this real quick. So what we're looking at here is a digital fader for levels. And 0 dB means the signal is not affected and we are

listening to raw sound, depending on the software that you use, the meter will look slightly different. But usually 0db is the default level.

But the software gives you a little bit of extra headroom and it lets you go up to 6db. If you go above that, your audio signals will start clipping. So its important that your goal should always be not to hit 0db.



Everything under 0 is a negative number. And we want our kick to be hitting around -10. If you feel it is quite, please turn up your speakers or headphone volume, because this is a perfect level where a kick should be.

Always keep that in mind that you should never touch the master channel fader, it should be perfectly at 0db and also you should never drop any

plugins on the master channel. That's the best way to do it, because if I mess with the Master Channel that I'm just kind of lying to myself about what I'm really hearing.

After setting up the kick you should use your ear, and level everything taking the reference of the kick level and make sure that the levels almost fit around this kick. This is going to get me in the ballpark. Remember we are doing this to get a good starting point and I'm pretty sure by doing this we will achieve it.

## 2. PINK NOISE METHOD

What you do here is instead of using instead of picking something from your project, you use pink noise as a reference point. Let see how to do it

- Step 1:- Create a new track and label it 'NOISE'
- Step 2:- Add a Pink Noise track to the track ([Download it From Here](#)) The Noise file that you'll use should be at -12db. If it's not, use the level fader and adjust it.
- Step 3:- Loop it all the way through the whole track.
- Step 4:- Pull down the levels of all tracks to 0db. Except the track with Pink Noise, bring down all the tracks to 0db.



- Take pink noise as a reference to level all the tracks. For eg. Take a kick and play it with the pink noise. While playing, bring up the kick level until you barely hear it. Then mute the kick leaving the level knob at that place where you were able to hear it.
- After Balancing the Kick, mute it and now play another track with pink noise then again bring it to a certain volume at which you can barely hear it, repeat this with all the track.

In the end, you will have all your tracks perfectly balanced, and now ready to be mixed

Once you do that with all those tracks, delete your pink noise and your mix should be with a reasonably good starting point.

But It's not the endpoint, though. That's not going to get you a perfect mix. This is to help you to get a good starting point with a well-balanced mix.

Now we will learn in detail about all the fundamental elements of mixing.





**PANNING**

08

# WHAT EXACTLY IS PANNING?

Ultimately, what panning does is help us to place the sound across the stereo field. Stereo field means where we are listening. The stereo field consists of placement from left to right and front to back. When you mix a song, you can set your instruments wherever you want them on the “stage” that’s created by your listeners’ speakers.

Let’s understand this with a diagram.



If we are sitting between these two speakers we are experiencing stereo sound, remember when we hear the term stereo, we’re talking about two speakers. So by using panning, we can make the sound feel like it’s coming from the right speaker or left speaker by panning it all the way to the right or left.

This makes the sound feel like it’s everywhere

between the two speakers and we can pan it even to the center and make it feel like it's coming from the area which is in the center of the speakers. This is called the Phantom Center. The way we do that is if we put an equal amount of a sound in the left and the right speaker, our brain is gonna make it feel like the sound is coming from the center.

Now there are some tricks you can also do to get even wider than the stereo field and make it feel like it's coming from a little wider than the speaker area. Without doing those tricks you're pretty much limited to the left and right speakers and everything in between. That's what panning is. We can use panning to add some movement to music, or we can make a synthesizer feel like it's moving back and forth quickly. We can also deal with that problem that we just talked about, that is conflicting instruments.

Let's say we've got three singers, a lead singer, and two background singers. Now the stereo field can work a lot like the visual field, if you're standing at the center of the stage and you're listening, to those three singers who are just standing on a straight line facing you then you may not be able to see those

other two singers because they're behind this first singer.

The same works in DAW when we align all the tracks to the center, it becomes difficult to make the song sound pleasing and it sometimes becomes harsh. So what we might do is pan one background vocals to a little bit left and another one to a little right, keeping the lead vocal in the center.

Here's a quick representation of how to pan instruments for wider sound:-



Beneath are some rules of thumb I follow when panning my tracks, with some additional information on how to make the most of the stereo field.

Let's start with how I generally pan instruments.

## OVERHEADS & ROOMS

I always pay close attention to the perspective of the drums. Sometimes audience perspective works, and sometimes it needs to be from the drummers' vantage point. It all depends on the interplay of the drums with the other elements, and how the arrangement was built from the ground up.

As a recording engineer, I'll record both mono and stereo overheads and room microphones, as I like having options when it reaches the mix stage. It's great to keep the drums more centered during verses, and bring in wide stereo overheads and rooms during the choruses.

Generally, I pan these stereo tracks pretty wide, usually 75% to the left and right. I pay special attention to polarity and phase relationships, and to how the different frequencies of these stereo drum recordings interact with the other elements.

## KICK & SNARE

I prefer Kicks straight up the middle. I do make an effort to keep elements with large amounts of low-frequency content away from the sides, so if I do place a roomy, stereo kick sample into the mix.

Certain styles of music (think EDM or Pop) use more “sound design-ey” kick and snare drums that were perhaps partially synthesized, or are a collage of multiple layers — and these designed drums might even have some side-to-side movement or stereo reverb effects baked into the sample. In this case, I call it by ear — if the drums work with the rest of the instrumentation, I leave them as is. If not, I’ll narrow their stereo width or make them mono entirely.

## HI-HATS

I sort of despise hi-hats, especially open ones. They are usually overly bright, noisy, and generally desperate for attention. If the stereo field of the drums is well established by the overheads and/or room, I’ll even keep the direct hi-hat level low, or mute the signal entirely. I do love dance-y 16th-note hats as long as they’re grooving correctly. Hi-hats are panned depending on how the perspective of the drums is set, and I usually keep them between 60-80% to the side.

Electronic hats are a bit different — some stay close to the center, some are made to be stereo, some are hard-panned, and some are subject to auto-panning and can move around the stereo field.



## BASS

Typically, electric bass stays straight up the middle, but sometimes a bit of subtle roomy width can add much-needed depth to a bass (just as it can kick drum) so from time to time I'll use a reverb plugin (or re-amping) to create a “bass room” track. This track will likely stay low in the mix, and may only be used during certain sections for spatial emphasis.

Synth bass is another beast altogether — subby, warm-sounding synth basses I'll likely keep up the middle, but for ripping, biting bass synths, sometimes keeping them wide is the way to go, especially if they were designed to have unique stereo properties.

## GUITARS

Having been an audio production and sound design professor for the greater part of the last decade, I can't tell you how many mix critiques I've given for student mixes that have doubled, distorted electric guitars panned straight up the middle. How exactly is the vocal supposed to thrive, let alone exist, amongst such chaos?

Double-tracked guitars are panned 100% in my world.



Try to balance it against other elements like keyboard, synths, horns, or even hi-hat. If there's a single stereo-tracked acoustic, I determine the width by ear. I may pull the stereo field in a bit, and spread it out during choruses or moments that are meant to be bigger.

Lead guitars tend to be very close to the center (if not dead-center) unless they are doubled in which case I'll start at about 10% panned apart, and experiment with different widths until I'm satisfied.

## **SYNTHS & KEYS**

There are so many variables here due to the versatile sonic nature of synthesizers and keyboard instruments. On a recent mix, I was committed to featuring a synthesizer part by keeping it stereo, but it was simply too busy and ended up overshadowing the vocal. I ended up panning it mostly to the right, and all of a sudden the entire song came to life.

I suppose the key is to manage the panning of your midrange instruments with a lot of care. Certain pads and organs benefit from being stereo, but it depends on what else is happening. Stereo-miked pianos are a beautiful sound on their own but can benefit from not being spread 100%, or if

emphasized a bit to one side when placed in the context of a mix.

I regularly use an auto-panner (Soundtoys Panman) to give organs or pads a bit of movement and to allow the lead vocal to really own the center.

## **HORNS, WOODWINDS & STRINGS**

For arrangements that include orchestral elements such as these, I reference a traditional orchestra seating chart and then consider if panning them in that manner is for the benefit of the mix. If it's a reggae tune with horns, I'll lay them out as the band would be on stage, again considering if my mix is from the audience or musicians' perspective. If they are solo elements but are present throughout the course of the song, I pan them considering how their timbre relates to other components of the instrumentation.

For example — if I have a solo viola, I'll try to keep it away from the acoustic guitar, and solo trumpet I try to keep apart from hi-hats, etc.

## **VOCALS**

There are so many ways to approach vocal

production, but one consistent thing in my mixing is that if there's a lead vocal, it's going to be right up the middle. An exception would be if there was a double-tracked lead, sometimes I may pan them each a bit symmetrically off-center. Ad-libs may also be a touch off-center, if not panned significantly.

My approach to panning backup vocals isn't so simple— with pop sessions, it's common to have dozens of layers of vocal tracks to work with, and so I'll place these layers all across the stereo field.

A technique I often use is to keep backup vocals a touch shy of 100% wide during verses and spread them out entirely during the choruses.

## **REVERB & DELAY**

I use both mono and stereo time-based effects on vocals. Similar to my approach to backup vocals, transitioning from mono or not-entirely-wide reverbs and/or delays during verses, to fully-wide effects in choruses can add a satisfying contrast to your mixes.

Vocal delay, in particular, can cloud up your mixes if you don't thoughtfully distribute them across the stereo spectrum. I'll decide different widths for

delays that have different note values (eighth, quarter, half-note, etc.) in an effort to create separation and maintain clarity in my mixes. One of my favorite techniques for epic, power-ballad individual tambourine hits is to pan the recorded tambourine to one side, send it through a reverb and pan that signal to the opposite side. I then time the pre-delay to taste (usually something similar to a sixteenth or eighth-note works best).

## SUMMARY

How I place elements in the stereo field is, of course, dependent on the arrangement and what I believe the song needs. As I mentioned, I'll also change how elements are panned throughout the course of the song. Additionally, I'll use a combination of auto-panners, doublers, mid-side processing, and wideners to maximize my usage of the stereo spectrum for the benefit of the mix.

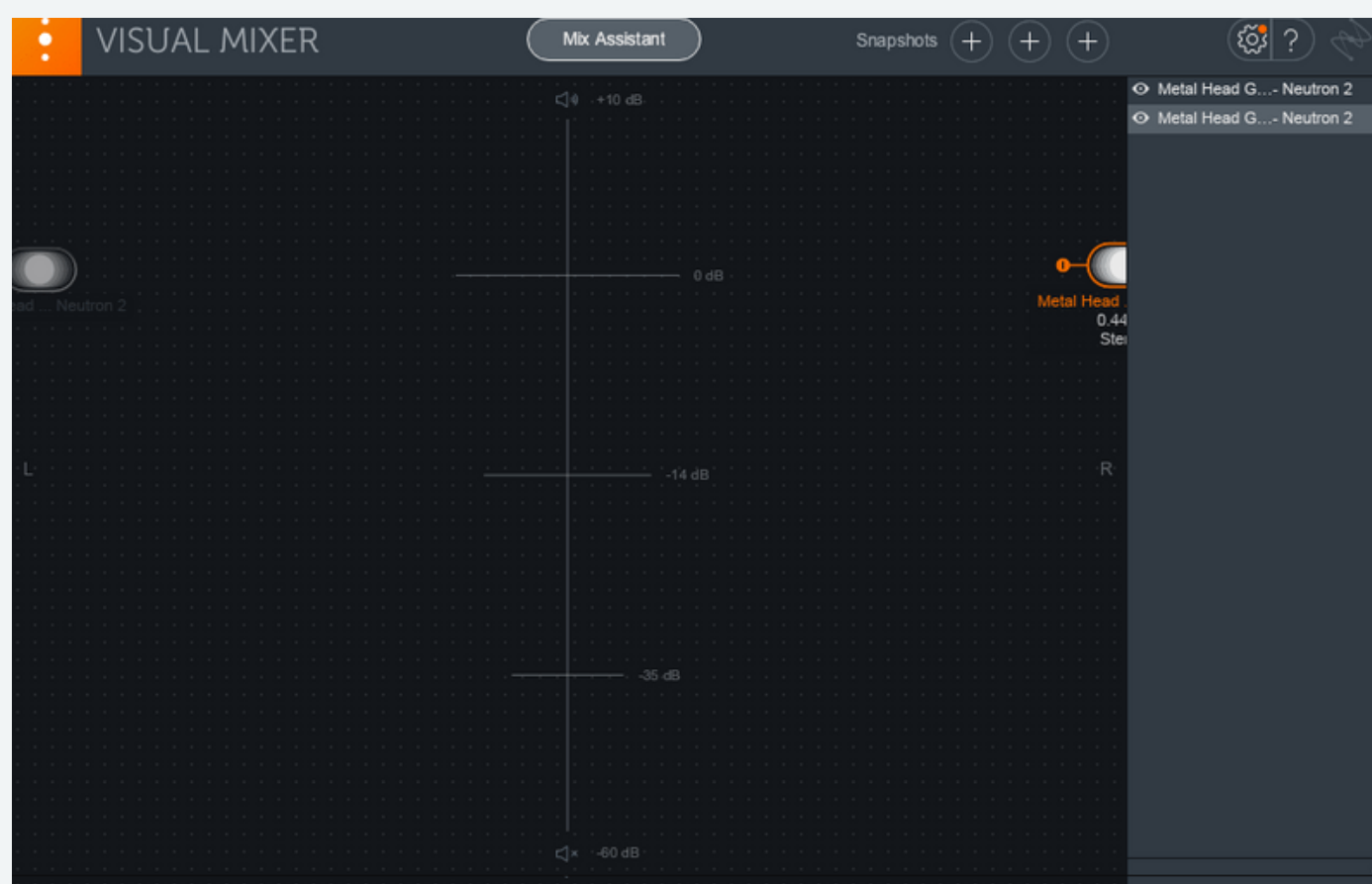
## MY BEST 11 PANNING TIPS

There are no hard and fast rules for panning, still, these tips will help you achieve a wide, full-sounding mix quickly.

## 1. TRY HARD-PANNING DOUBLED INSTRUMENTS

If you have double-tracked instruments (the same part recorded twice, for effect), try panning them hard left and hard right. This is a classic move with heavy electric guitars, though it can also work with instruments such as EDM synths (think drops) and saxophones (think funk). The effect is a fuller-sounding mix.

The difference is staggering. All we changed was the panning scheme, as shown here in iZotope's Visual Mixer.

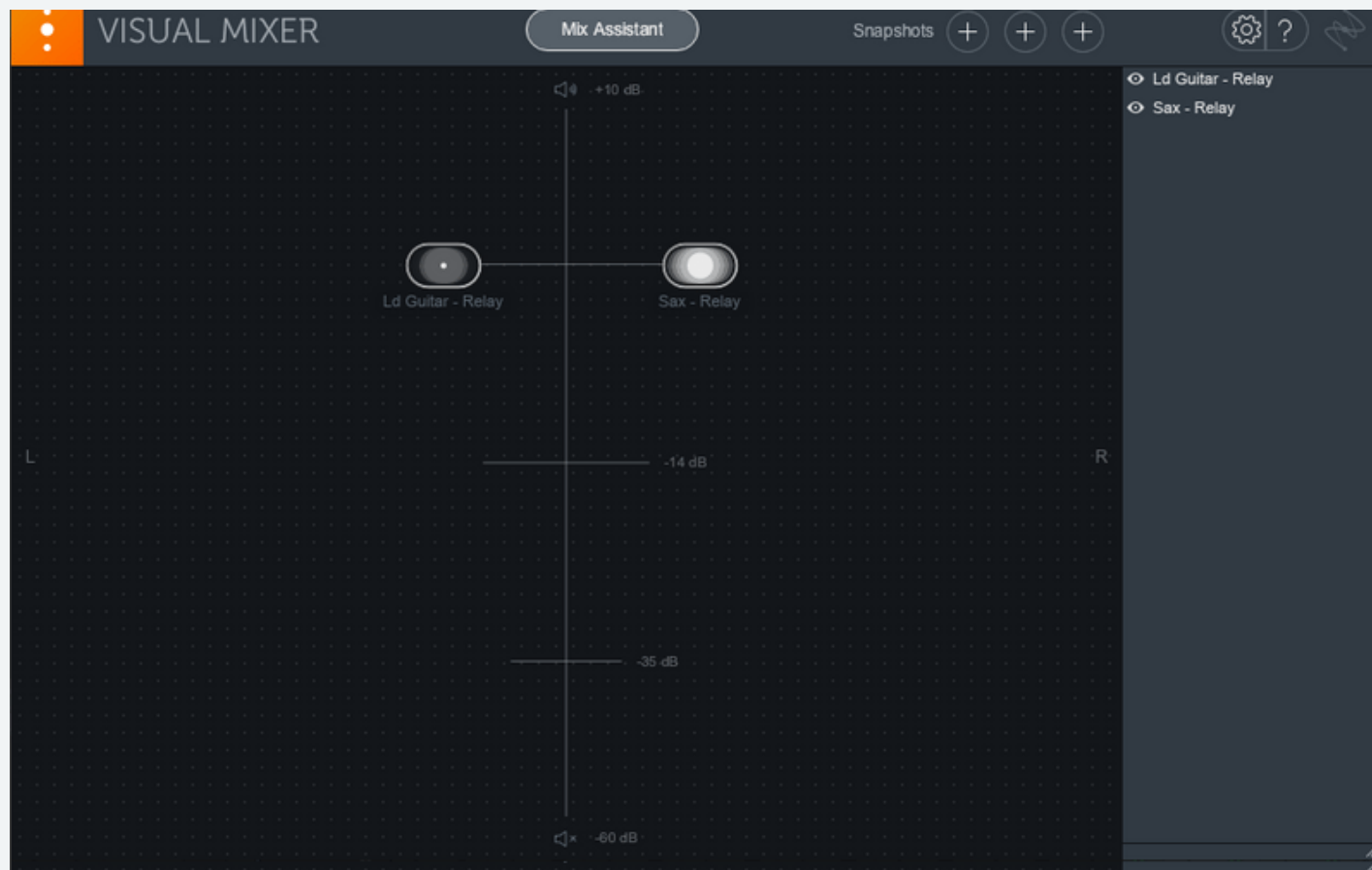


## 2. COMPLEMENTARY PANNING

If you have two instruments occupying similar frequencies, try panning them opposite of one another, say 20 percent left and 20 percent right.



I recently did this for a series of jazz tunes, where an electric guitar doubled the saxophone during the melodies. Panned in the center, both parts clashed. Panned slightly to the sides, they sounded correct.



Don't go hard left or hard right with this trick. A guitar panned slightly to the left will better complement a keyboard panned slightly to the right, creating a more balanced feel.

Remember: complementary panning will invite a listener into the sound, whereas blasting two rubbing instruments from the same spatial location will be more confrontational. Sometimes that's okay but you should always know what you're doing.

Snare drums usually appear in the center image, even though they're not centered on a drum kit. The same is true for the bass guitar and the kick drum. Put these elements anywhere but the center, and you're immediately calling attention to them in an eccentric, offhand way.

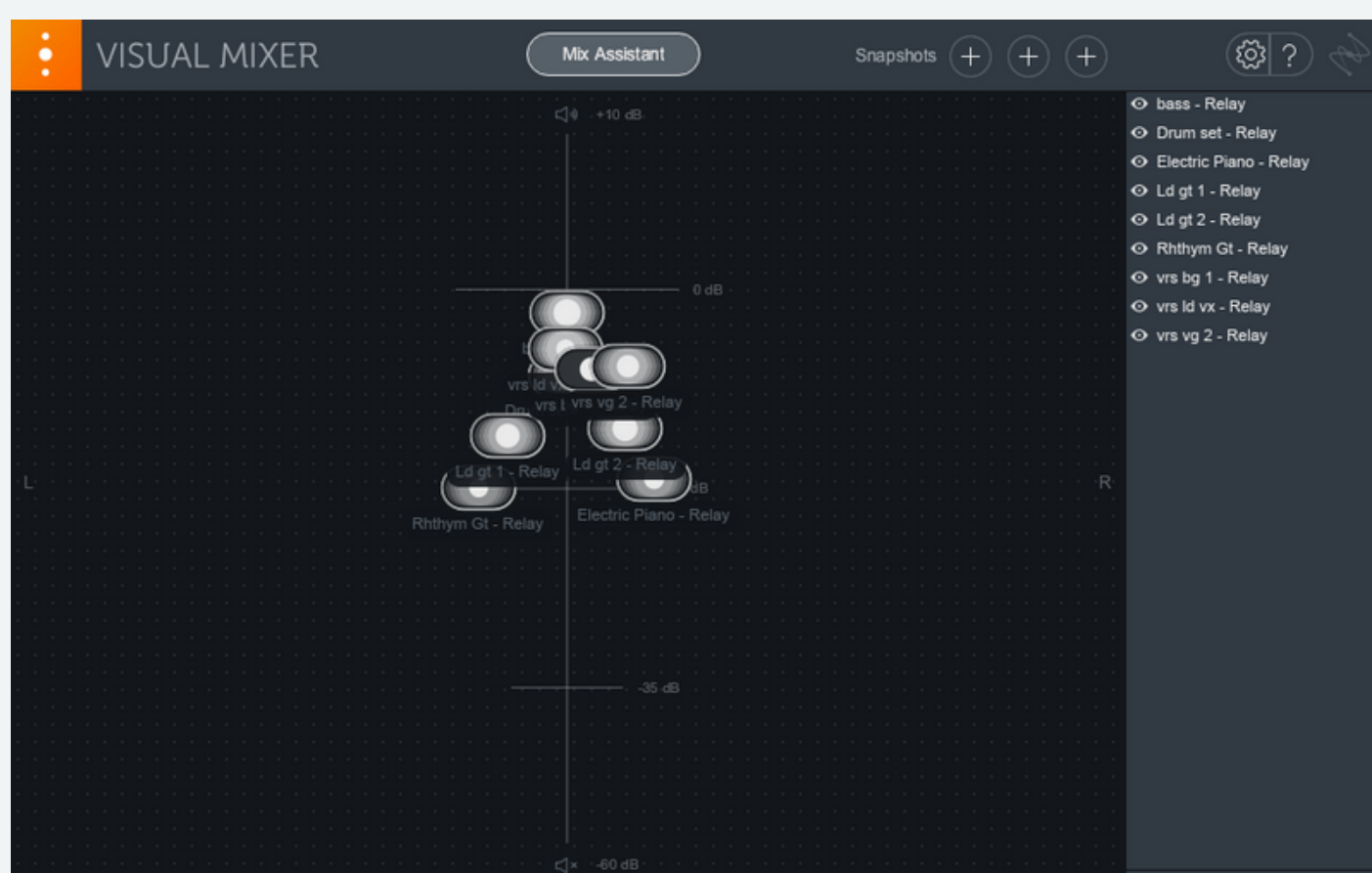
This, of course, could be just the ticket. It's well within your rights to mess with tradition. But you must first know that it's proper to do so—and the only way to know is to pay strict attention to conventions.

## **4. TRY NARROW VERSES AND WIDER CHORUSES**

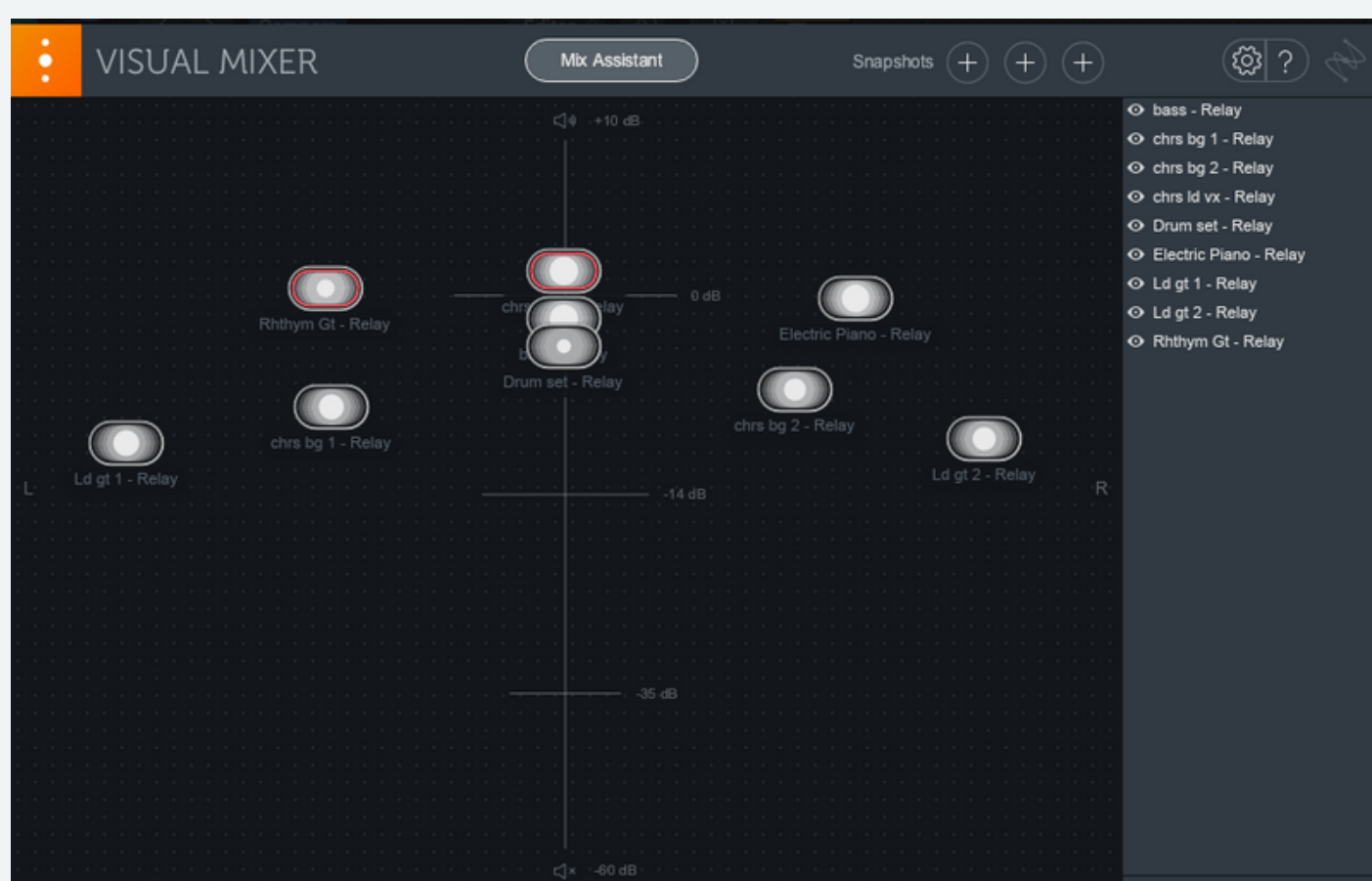
In today's world, we don't get a lot of dynamic range. People still want things loud, loud, and more loud—despite the loudness wars being supposedly over. One of the ways we can create the illusion of dynamics is through creating different panning schemes for the verse and chorus.

During the verses, try a narrower panning scheme, something like this:





Then, when it's time for the chorus, move the instruments wider, like so:



This will create a similar feeling of arrival, making the mix feel louder without necessarily becoming louder.

## 5. CHECK YOUR WORK IN MONO

Always check your mix in mono to ensure you aren't losing too much in the fold-down. It's possible to spend a long time panning everything, only to go too far and realize your mix sounded more impactful

before you even began.

If you're taking a single instrument and using delay tricks or polarity manipulation to achieve width, you must monitor the changes of this effect in mono. Each part may sound more distinct, in stereo, with some polarity trickery on the guitars making them feel wider.

But now listen in mono:  
It's barely there!

Unfortunately, mono compatibility is still a thing. It's important to club mixes, to supermarket broadcasts, and more. Always keep it in mind.

## **6. CONSIDER THE CLUB**

If you're mixing any form of electronic music likely to be played in a club setting, bear in mind that many playback systems are mono. And, if they're not wired to be mono, they can often feel functionally mono, because of how sound travels in big packed spaces.

What does this mean for you? Pay special attention to elements such as reverb, panning moves, and more. Your fancy trickery on a transitional riser might

lose all impact in a club.

Check in mono, as stated above. I also find it's helpful to check club-style mixes through the impulse responses of big, full clubs. You can use fancy headphone modeling software for that, or download free impulse responses of clubs from sources such as this.

Load up the impulse response in your favorite convolution reverb, and just listen from time to time to make sure the mix sounds as good as a reference, also pumped through the club IR. It's not a fool-proof solution, but it helps.

## **7. HOWEVER YOU PAN, CHECK YOUR MIX IN VARIOUS SOUND SYSTEMS**

Translation checks are always a good idea—especially so when it comes to panning decisions. Check your mix in headphones to make sure the mix doesn't sound disjointed or off-balance. Your monitors might be excellent, but headphones sound different, as they eliminate crosstalk (the phenomenon of information from the right speaker reaching the left ear and vice versa).

Not only do headphones sound different—they're also everywhere: I'll bet more people listen over headphones than stereo systems nowadays.

Also, check your mix car stereos if possible, as no listeners really sit in the center position of a car stereo; be sure your panning decisions make sense if you're sitting a little to the left or right. It's ultimately a polite thing to do for the listener.

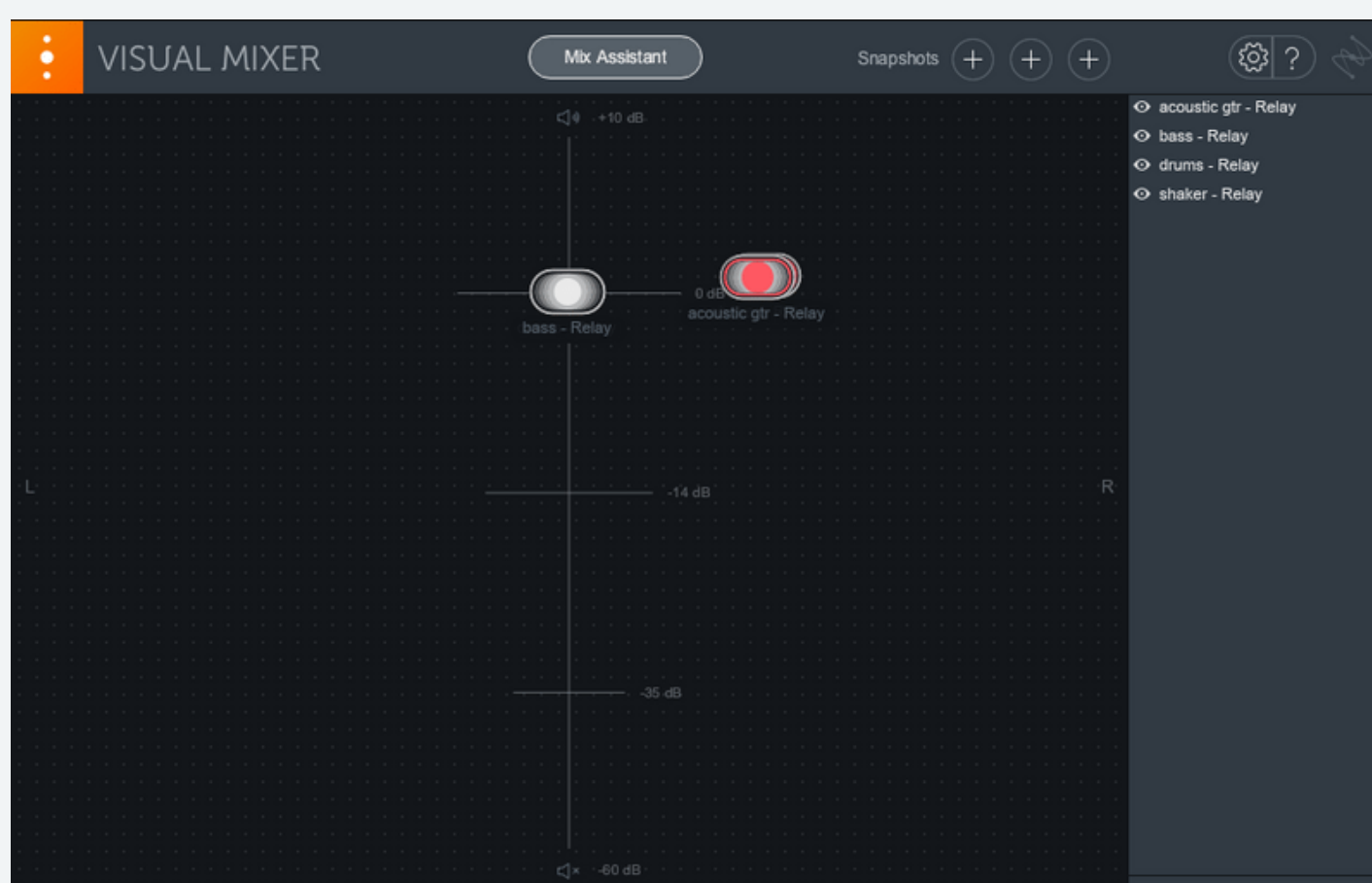
## **8. DON'T OVERDO ANY ONE PANNING LOCATION UNLESS YOU WANT TO STRESS OUT THE LISTENER**

Panning is a way of achieving balance, just like frequency manipulation and dynamic control. If you choose an arbitrary location within the stereo field and jam it chock full of trebly, busy instruments, you're going to tax the listener. I recommend not taxing the listener unless you're going for a specific effect, such as building tension to an inevitable release (this is where the use of complementary panning, after the tension build-up) would come into play.

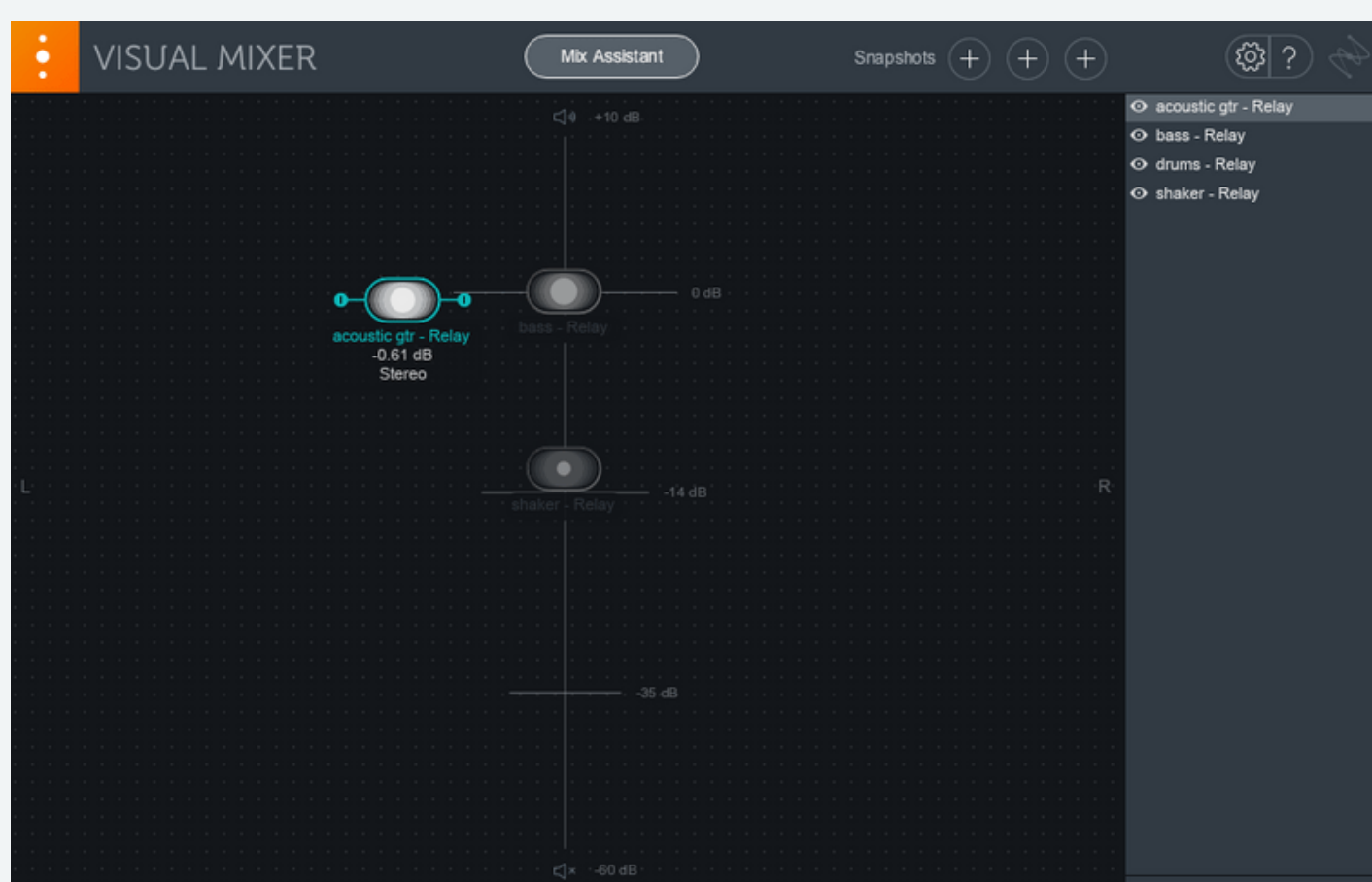
Take the example of an acoustic guitar and a hi-hat. Both are usually marking straight subdivisions in a

mix (8th notes or 16th notes). Both have a lot of high-end information, and both can be jangly or harsh.

Here's an acoustic guitar, a drum set playing a hi-hat pattern, and a shaker, all in the same mix. The hi-hat, guitar, and shaker all panned to roughly the same spot, like so:



It sounds pretty harsh to me. But if I move these three elements out of each other's way, I achieve a better balance.





## 9. USE PANNING TO EVOKE CLASSIC GENRES

Sometimes older recordings, or modern recordings mixed to evoke a nostalgic vibe, employ drastically weird panning schemes—such as placing all the drums in one speaker, and the rest of the band in another. This can be used for effect. Rilo Kiley and Elliott Smith both utilized this technique in their later work to achieve specific textures.

## 10. LESS IS MORE

Sometimes the widest sounding mixes don't come from panning everything. They come from panning just a couple of interesting elements while maintaining a strong and balanced center.

Try just making just one element of your mix wide and spacious, like doubled-guitars, a stereo piano track, or drum overheads, and make everything else work around the center with careful level setting and judicious EQ. You'll be surprised how powerful this can be.



# EQUILISATION



Now, this is probably the most tedious of all of them and also debatably, the most important.

The EQ is kind of our scalpel when it comes to fixing a mix. This is the most important tool that we can use to make our music both sound better and cleaner.

It is simply a tool that lets you adjust the volume of the individual frequencies within an audio source. Rather than a volume fader, which would allow us to adjust the overall volume, an equalizer allows us to just turn up or turn down individual frequencies and individual elements of that sound.



Every instrument has a fundamental note. Along with a fundamental note, it has overtones. That's what gives an instrument its tone or its character, its timbre, and that's why a bass guitar,

for example, sounds different than an organ.

Whenever you're using an EQ. You can do 4 things,

- **Remove nasty elements**

1. Narrow cuts to remove room resonances.
2. High pass filter (only if needed) to remove low-end noise.
3. Do this in solo as part of the preparation phase.

- **Enhance pleasing elements**

1. Wider cuts and boosts to shape the tone.
2. Be bold if necessary - but only if you know what you are doing.

- **Make things sound different**

1. For example, filtering all the lows and highs to create the 'telephone' sound on vocals.

- **Create space in the mix using range allocation**

1. Don't boost two different channels in the same frequency range.
2. Instead, carve out space for important parts e.g. vocals or lead guitar.

We will discuss about this in detail, but first understand that by adjusting the frequencies, by

cutting certain frequencies or boosting others, we can adjust that tone and change the timbre of the instrument.

Now, it's important to bear in mind that you can't completely change the sound of an instrument with EQ alone. All you can do is work with what's already there.

In the recording phase, you decide what tone you want, and then you use EQ to scope that and make small changes to take it further towards your end goal.

## HOW TO USE AN EQUALIZER

Before you can truly learn how to use EQ, you need to understand the frequency spectrum. You could use an EQ chart for this, but let's just take a look at an EQ plugin instead



We go from left to right, and we start with 20 hertz. Then on the right, we've got 20,000 hertz or 20 kilohertz. This is the range of human hearing.

Now, none of us will actually be able to hear 20 kilohertz. When you're first born you can, but as you get older, your ability to hear all frequencies slowly degrades.

But you will still hear the impact of 20 kilohertz – so don't ignore it.

Bass is on the low end (left). You can feel 20 hertz if you're on a really large sound system – but not necessarily hear it.



In between these two extremes, we've got the human range of hearing.

For me, this breaks down into 6 very distinct



sections.

## **SUB-BASS (20-60HZ)**

The first area we're going to focus on is sub-bass.

Everything below 60Hz is sub-bass, so generally you need a subwoofer or a good pair of headphones (open-back headphones, for example) to hear that.

You should be able to hear it a little bit if you're on monitors or headphones. But if you're listening on a laptop or a phone, there's no way you will hear that.

## **BASS (60-200HZ)**

After that, we get into what I would call bass. For me, this is everything between 60 and 200 hertz.

In this area, we've got lots of bass guitar. Lots of the low-end vocals as well, because male vocals are going to have the fundamental below 200Hz in most cases.

## **LOW MIDS (200-600HZ)**

Next up, if you go from 200 up to 600 hertz, this is what I would call low mids, and this is a really important area for mixing.

Now, this area is crucial for home recording, because this is where you get a lot of buildup with guitars, vocals, even the top end of the bass guitar especially.

This is an area that's really guilty for adding mud to a mix.

## **MIDS (600HZ-3KHZ)**

The human hearing focuses mostly on this frequency range.

So it's crucial to get this range right. You want the main focus of the track (e.g. vocals) to have lots of room in this range.

Be aware that this is also where you can start to get into harshness and aggressive tones.

## **UPPER MIDS (3-8KHZ)**

Then we've got upper mids between 3 and 8 kilohertz, and this is where things really start to get harsh. This is where we have brittleness a lot of the time.

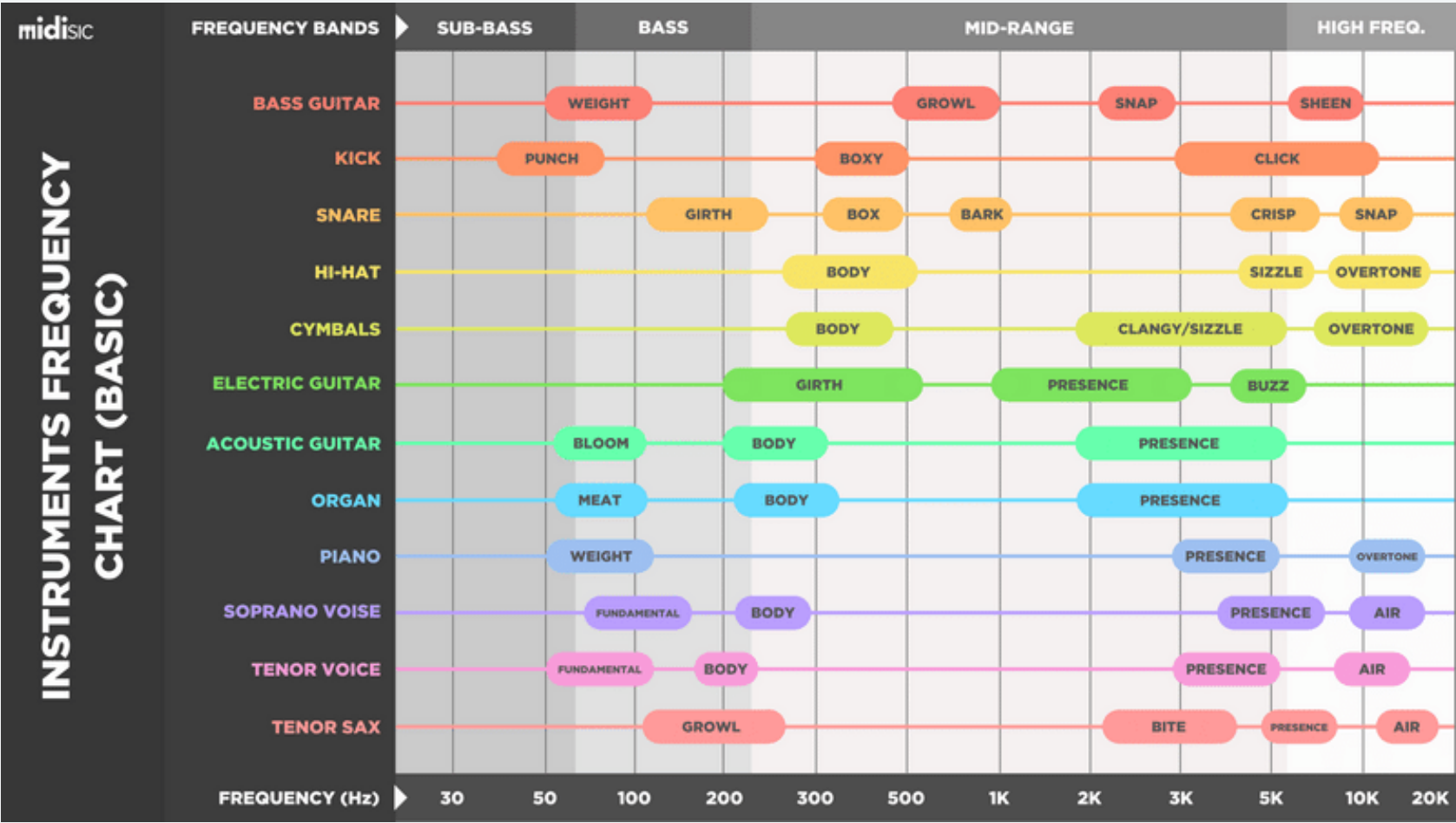
It's also an important range for clarity and

aggression, especially in vocals.

# HIGHS (8KHZ+)

After that we get to treble, or the highs. This is everything above 8 kilohertz. This is where we have air.

You could split this even further into 8-12kHz, and that’s what I would call treble, and then 12kHz+ is what I would call air. But for now, we’re just going to leave this as the highs, and this is everything above 8kHz.



So there you go. That’s a breakdown of the entire frequency spectrum. I recommend you learn this EQ mixing chart by heart.

Now, let’s learn about EQ’s.



EQ's are tools that a mixer can use to boost or cut any of these frequency ranges. They're the most frequently used tool in a mix.

Let's look at different types EQ:

## WHAT ARE THE DIFFERENT TYPES OF EQ AND FILTERS?

There are multiple variations of equalizers. Each has a different function, purpose, and characteristic sound. However, the most common types of EQ used in music production are parametric, semi-parametric, dynamic, graphic, and shelving.

Here are the most common types of EQ used in music production:

## PARAMETRIC EQ



Parametric equalizers are the most common and versatile type of EQ used in music production. These multiband equalizers offer fully configurable and adjustable frequency bands. You can control the center frequency, level, and bandwidth of each frequency band with the highest precision.

Common parameters found on a parametric EQ include:

**Frequency:**

Adjusts the center frequency range for each selected band. It also sets the frequency cutoff point for high and low cut filters.

**Resonance or Q:**

Adjusts the bandwidth of the selected band. It determines how wide or narrow to boost or cut frequencies. Higher Q values affect a narrower range of frequencies and lower Q values affect a wider range of frequencies. The Q means ‘quality factor.’

**Gain:**

Sets the gain amount for the selected band. It adjusts how much to cut or boost the center frequency. This parameter is often deactivated when

selecting the low cut or high cut filter shapes.

**Filter Slope:**

Sets the steepness of the filter when selecting either the low cut or high cut filter. Slope values range from 6dB to 96dB per octave.

**Filter Type:**

Allows you to choose the filter shape for the selected frequency band. Standard filter types are low cut, high cut, low shelf, high shelf, notch, and bell.

Parametric equalizers excel at shaping the tone and doing surgical work. You can cut harsh, unpleasant, or masking frequencies with pinpoint accuracy.

There are also digital and modeled analog style parametric equalizers. Digital style EQs like FabFilter's Pro-Q 3 are more advanced and provide modern features. Whereas analog modeled EQs like Waves' SSL G-Equalizer are more basic and have fewer bands or features. Moreover, digital style EQs are transparent sounding. They don't color the sound like an analog modeled EQ does when cutting and boosting frequencies. For instance, a common

technique is to cut with a transparent sounding digital style EQ and boost with an analog style EQ to add character.

In addition, parametric equalizers are great for creative processing. For example, creating filter effects, wobbles, adding movement, or shaping sounds in unique ways.

## SEMI-PARAMETRIC EQ



A semi-parametric equalizer is a parametric equalizer without one or more features. Typically, the bandwidth Q has a fixed setting. You can only adjust the frequency and gain of each band. Many also don't have an interactive display with an analyzer.

Semi-parametric equalizers are not as flexible as a fully parametric EQ. However, the fixed bandwidth



curves do an excellent job of boosting or attenuating a wide frequency range. They're ideal for tonal work and sweetening.

## DYNAMIC EQ



Dynamic equalizers combine the precision of parametric equalization with the dynamic control of compression or expansion. This versatile combination gives you more control over shaping and enhancing sounds.

What are the differences between parametric and dynamic equalizers? Parametric equalizers are linear processors, meaning they treat the incoming audio continuously. For example, filter cuts or boosts are static and will not change.

Dynamic equalizers are nonlinear processors. The

filters react to either the internal audio source or are triggered by a sidechain input source. For example, cuts and boosts respond to the incoming audio that passes a set threshold level. This dynamic movement adapts to the music.

## GRAPHIC EQ



Graphic equalizers boost or attenuate a range of fixed frequencies using a bank of evenly spaced slider controls. They can also have up to 31 or more bands. More bands will provide higher accuracy over the frequency spectrum.

Moreover, graphic equalizers with narrower bandwidths have greater precision. However, they're not as effective for surgical work as a parametric EQ. There is no control over the filter shape and bandwidth of each band. Instead, graphic equalizers are generally used to make broad changes to the



overall mix or bus groups.

When working with a graphic equalizer, it's better to make small, incremental adjustments across the frequency spectrum. You'll get smoother results by rounding out your mix. Avoid making drastic adjustment leaps between each frequency band. It will sound jarring and unnatural.

## SHELVING EQ



Shelving equalizers are the most basic type of EQ. A shelving equalizer boosts or attenuates frequencies above or below a specified cutoff point. Many shelving equalizers also have predetermined filter curves with a wide Q. Their smooth sloping curves excel at emphasizing or attenuating frequencies in a broad, musical manner to achieve more clarity and presence.

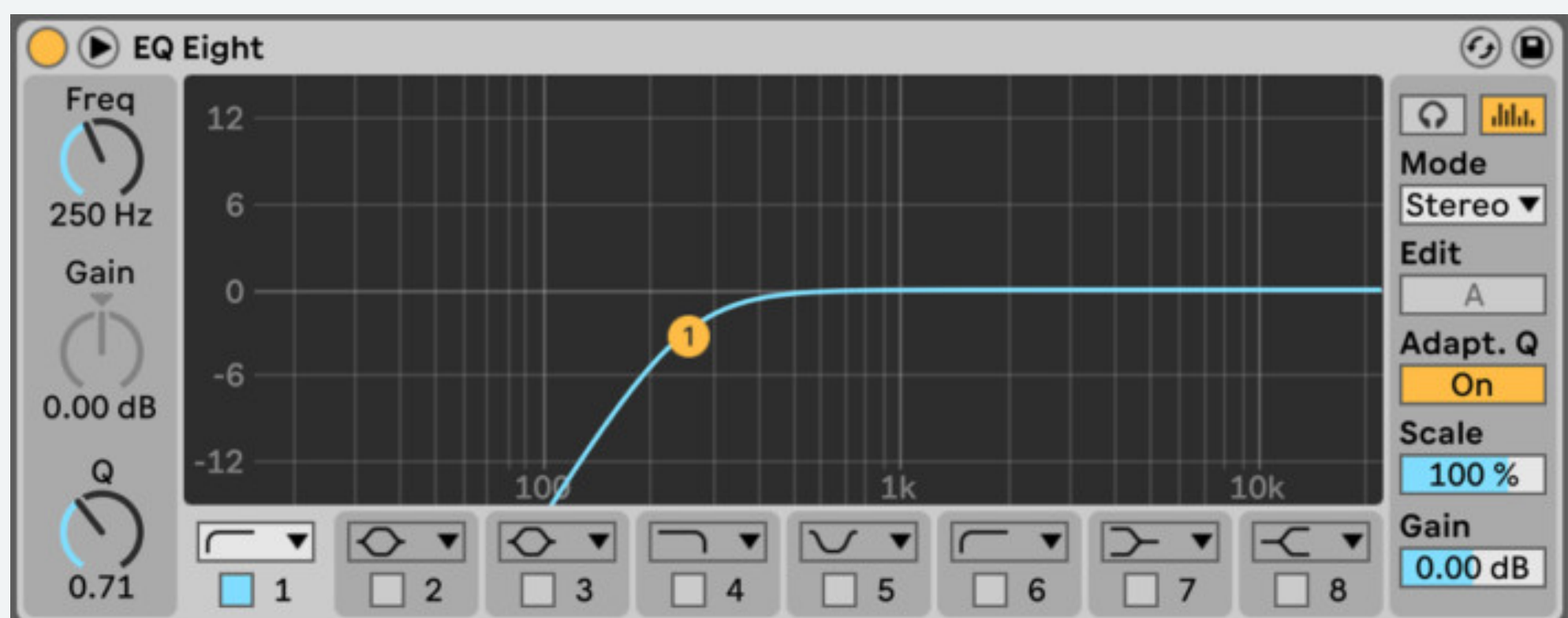
Most shelving equalizers have high and low shelving filters. Some also feature midrange bell curve filters.

They work excellent at sweetening and deepening full mixes. For example, use a shelving equalizer to create the familiar “smiling” EQ curve by boosting the high and low end of a mix.

## COMMON EQ FILTER TYPES

Understanding the different filter types and how they change the signal is also crucial. Below are the primary EQ filter types:

### LOW CUT FILTER

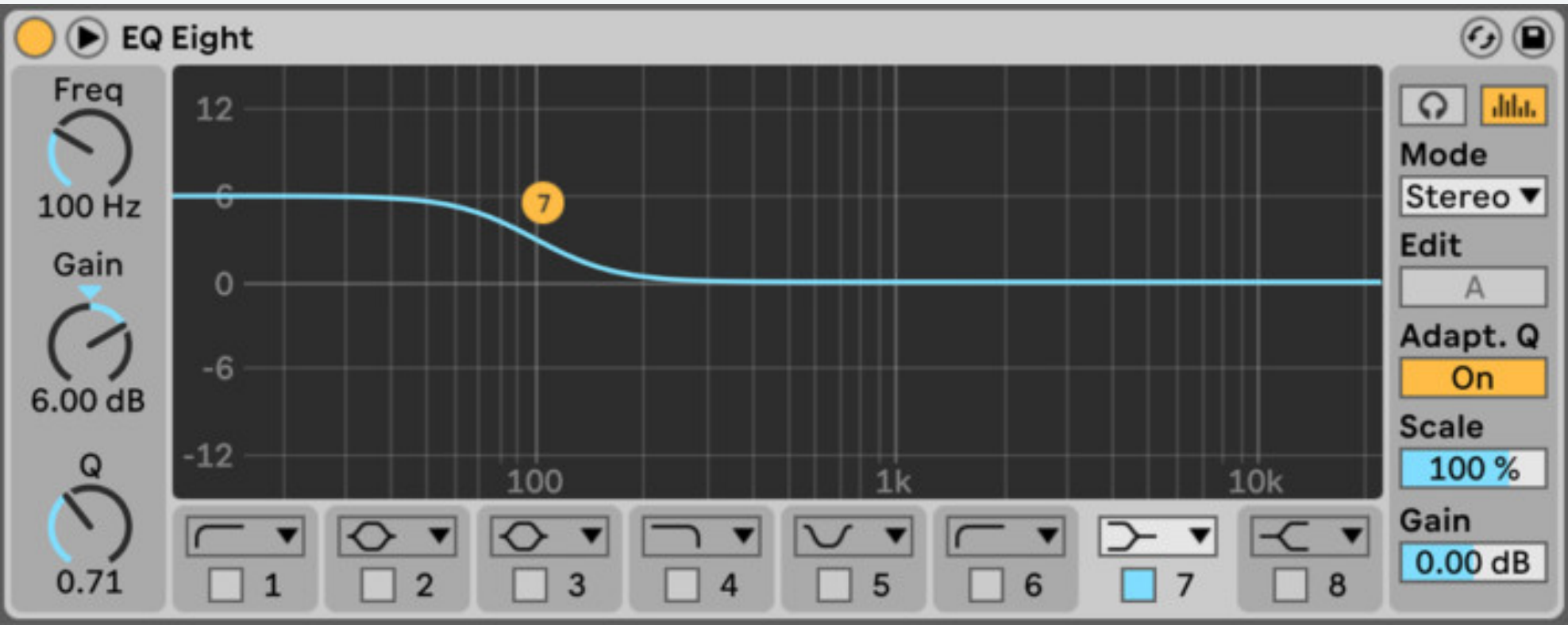


A low cut filter removes all frequencies below a specified frequency cutoff point. This filter is also known as a high pass filter because it passes all high frequencies above the cutoff point.

Use a low-cut filter to remove unwanted or problematic low frequencies. You can also automate

the frequency to create sweeping effects.

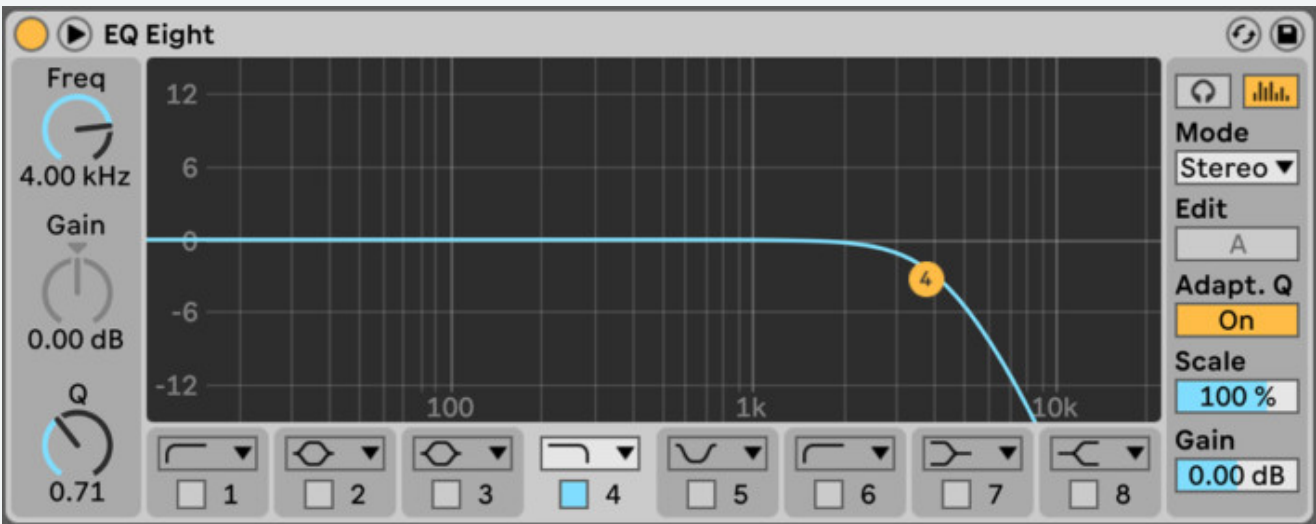
# LOW SHELF FILTER



A low shelf filter attenuates or boosts frequencies below a specified frequency point. Low-shelf filters don't cut frequencies out completely like low-cut filters. Instead, they gradually reduce or boost bass frequencies.

Use a shelf filter to boost or reduce low-end frequencies without cutting them off completely. It is useful for making broad tonal changes.

# HIGH CUT FILTER

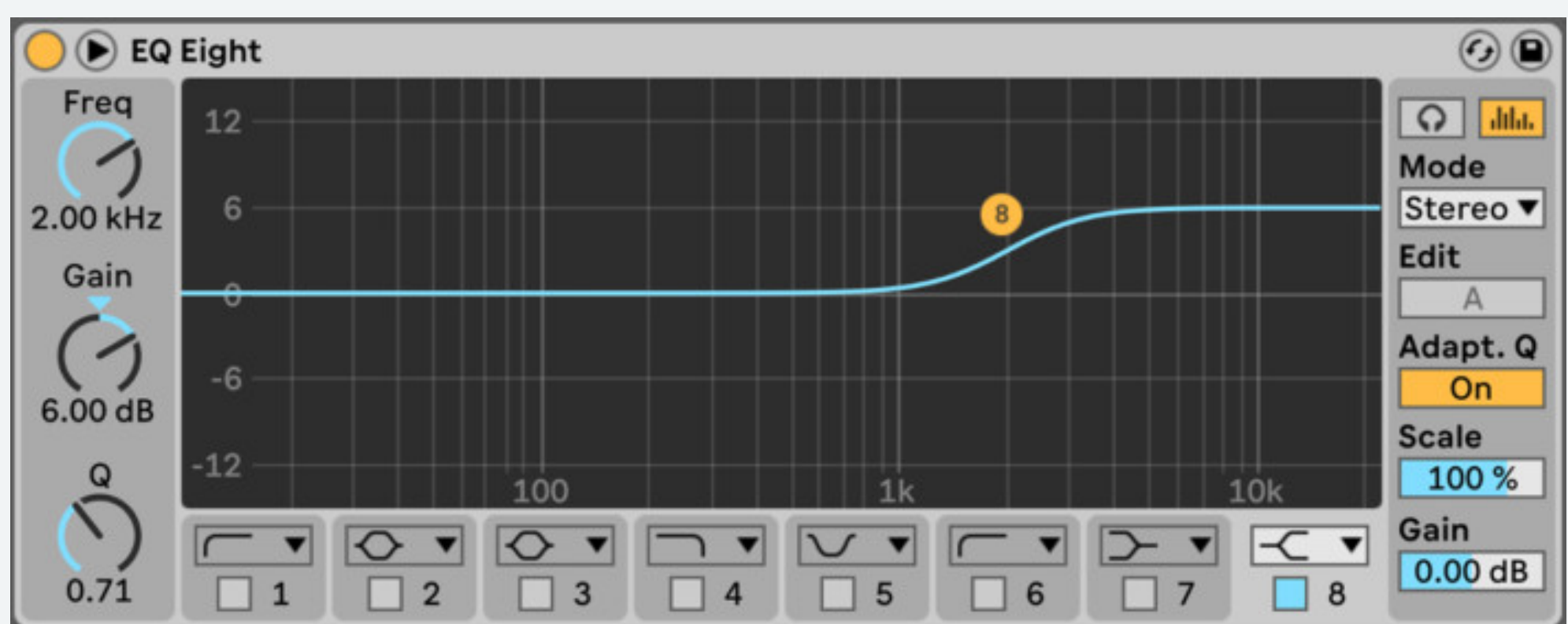




A high cut filter removes all frequencies above a specified frequency cutoff point. This filter is also known as a low pass filter because it passes all low frequencies below the cutoff point.

Use a high-cut filter to remove unwanted or problematic high frequencies. You can also automate the frequency to create sweeping effects.

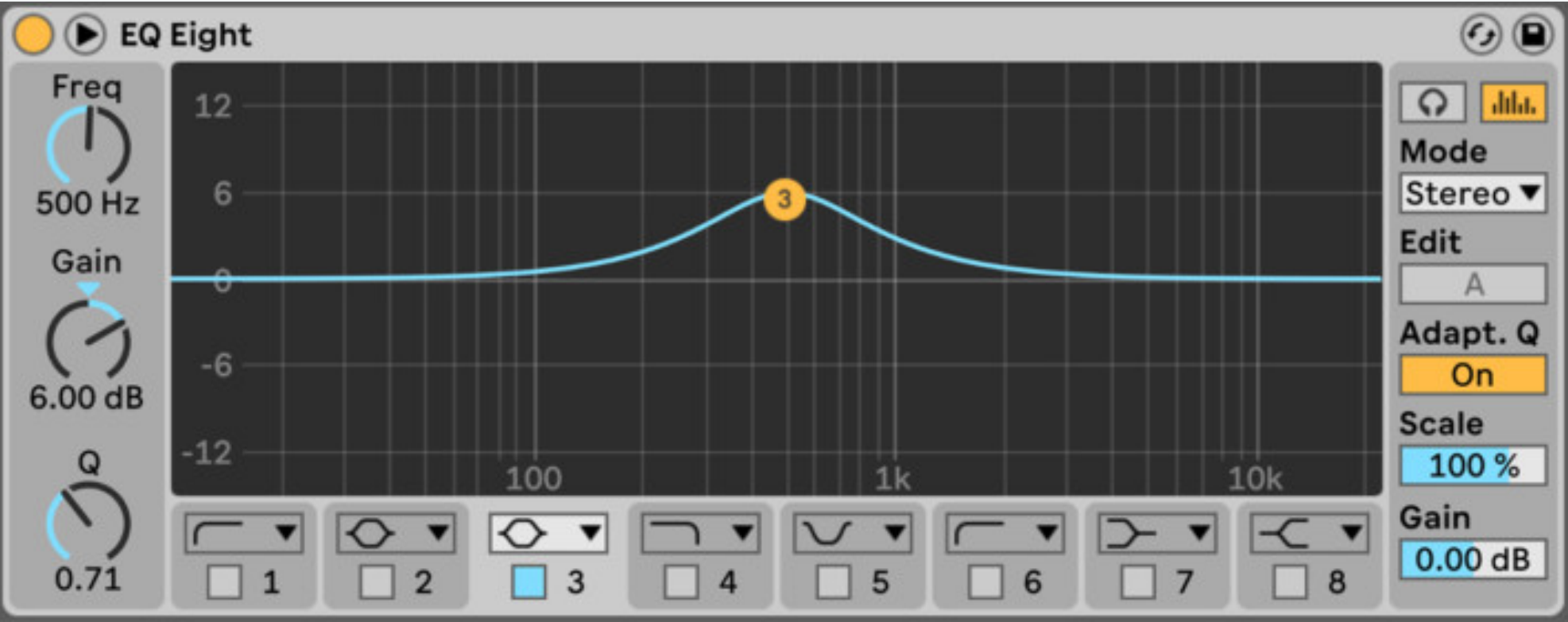
## HIGH SHELF FILTER



A high shelf filter attenuates or boosts frequencies above a specified frequency point. High-shelf filters don't cut frequencies out completely like high-cut filters. Instead, they gradually reduce or boost treble frequencies.

Use a shelf filter to boost or reduce high-end frequencies without cutting them off completely. It is useful for making broad tonal changes.

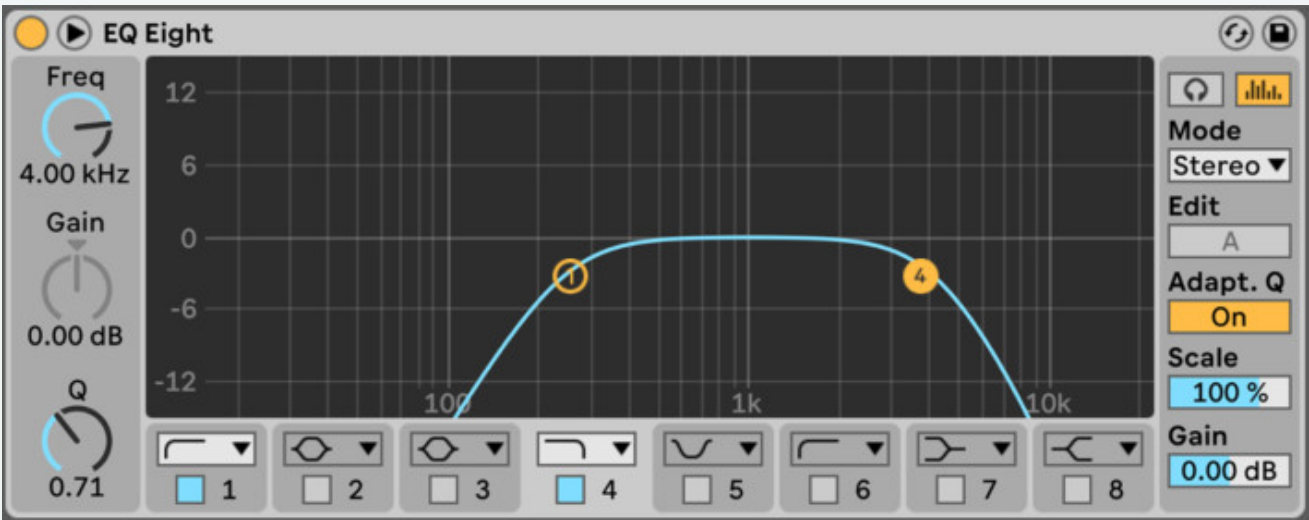
# BELL CURVE FILTER



A bell curve attenuates or boosts frequencies around a specified center frequency point. The bandwidth Q sets the width of the bell curve. Bell curve filters are also known as peak filters.

Moreover, bell curves are the most versatile filter. Use them to boost or reduce a range of frequencies with precision. For example, you can set broad curves to boost musical frequency areas or set narrow cuts to do corrective work with pinpoint accuracy.

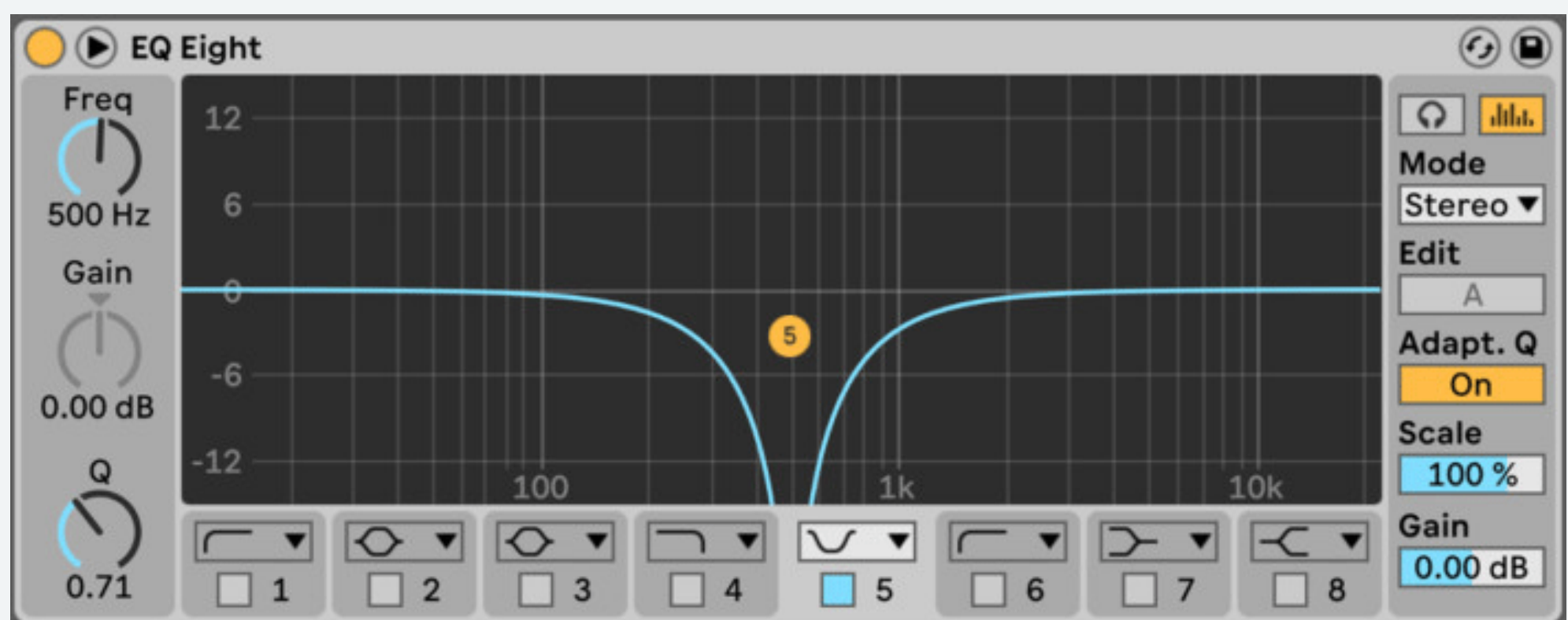
# BAND PASS FILTER



A band pass filter passes a range of frequencies around a specified center frequency point. Low and high frequencies outside the range are attenuated.

Use a band pass filter to isolate a range of frequencies. Multiband equalizers use multiple band pass filters to divide the audio spectrum into sections.

## NOTCH FILTER



A notch filter attenuates a range of frequencies around a specified center frequency point. Low and high frequencies pass on either side of the frequency range. Notch filters are also known as band stop filters.

Use a notch filter to cut a range of frequencies. Modulating notch filters also creates interesting tonal shaping effects.



# CONCLUSION

Equalization is critical for creating a polished mix and a professional sound. Knowing how to use the different types of EQ and filter shapes will help you make informed mixing decisions. Also, knowing what type of EQ to use for different situations will give you better results and speed up your workflow.

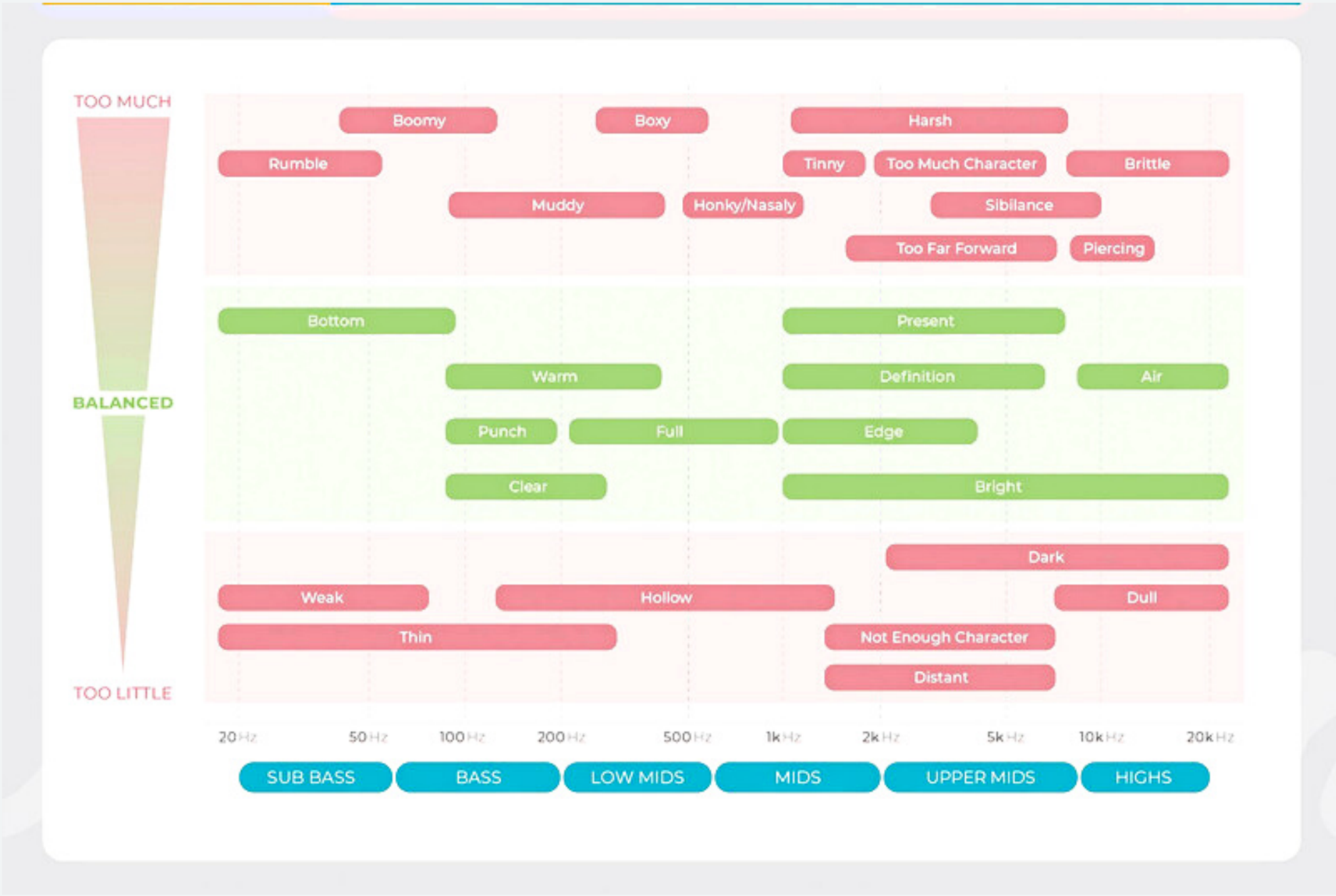
## HOW TO USE AN EQ CHART

EQ charts are a great way to learn to make intentional decisions when using an EQ.

Essentially, they break down the entire frequency spectrum and use descriptive words to explain each frequency range and how it sounds.

The point of an EQ is to balance the frequency spectrum. These charts are helpful when you're trying to figure out where the sound is unbalanced.

Learn these frequency charts by heart and you'll get significantly better at EQ. If you know something sounds brittle, that's going to be the upper mids; if you know something sounds muddy, it's going to be the lower mids, for example. It's easier to find the locations of the problems you hear.



While listening to an instrument, think: what does this sound like to me? If you're using green words, great. But if you're using any red words, it may be time to bring out an EQ.

It's also important to realize that EQ moves aren't black and white.

Sometimes, you might listen to an instrument and think it sounds thin (which, according to the chart, means it's lacking low end). So you'll reach for an EQ and boost the low end. But that doesn't help like you thought it would. It just makes the sound more cluttered.

What's going on?

Sometimes an instrument you think is lacking energy in one area actually has too much energy in another area, or vice versa. It makes the sound unbalanced.

So, if you boost the low end, and it sounds more cluttered, it may actually have too much energy in the upper mids. You make a cut around there, and all of a sudden the instrument sounds balanced.

The moral of the story: use your ears! If you think a particular mixing decision will sound good and it doesn't, don't keep it just because you think you should. Do some problem-solving. There's probably something else you could do to achieve the same result.

Now, a few warnings:

These charts are great for learning what different parts of the frequency spectrum sound like, but make sure to keep them from becoming a crutch. These are great learning tools, but shouldn't be referenced constantly.

Okay... now we're ready to dive into those 4 key approaches.

# FOUR KEY APPROACHES WHILE EQUING

Okay, we're getting somewhere. Time to talk about strategy.

Forget everything you know about EQ.

Let's keep it simple...

There are only four ways to approach EQ. Think of your equalizer as four completely different tools depending on how you use it.

## 1. REMOVE NASTY ELEMENTS

Let's go into the first approach, which is how most people start off.

This is using EQ to remove nasty elements. You can use narrow bands to remove nasty elements, called surgical EQ, which can really clean up any sound source BEFORE you've even done any tonal shaping.

By simply removing nasty elements, you make more room for the good stuff to come through.

What you're doing is surgically removing certain frequencies. It's surgical because you're using a



really narrow cut.

If you can remember earlier in this guide, I said about using narrow bands for cuts and broader bands for boosts, and that's exactly what you're doing here.

All you do is boost a narrow band and sweep around until you find a nasty sound. What you're listening for is a sudden increase in volume, because that suggests that there's lots of that frequency – which probably means it's a room resonance, because every room will have certain frequencies that resonate.



Once you find the frequency range, just cut it out by 2-10dB. I normally find one or two problematic ranges on important parts like vocals, guitars, snares, etc.

Generally, you want to avoid the solo button when mixing.

But when using this approach of removing nasty elements, it's okay to solo the channel. You can even do these surgical cuts BEFORE you start mixing, in the preparation phase.

This first approach also includes using high-pass filters to remove low end noise when necessary.

But don't go crazy with this – only use high-pass filters when you notice low end noise that needs removing, or you have another specific intention (like tightening up the bottom on a bass guitar).

Don't just use high-pass filters on everything, otherwise, your mix could end up sounding thin and weak.

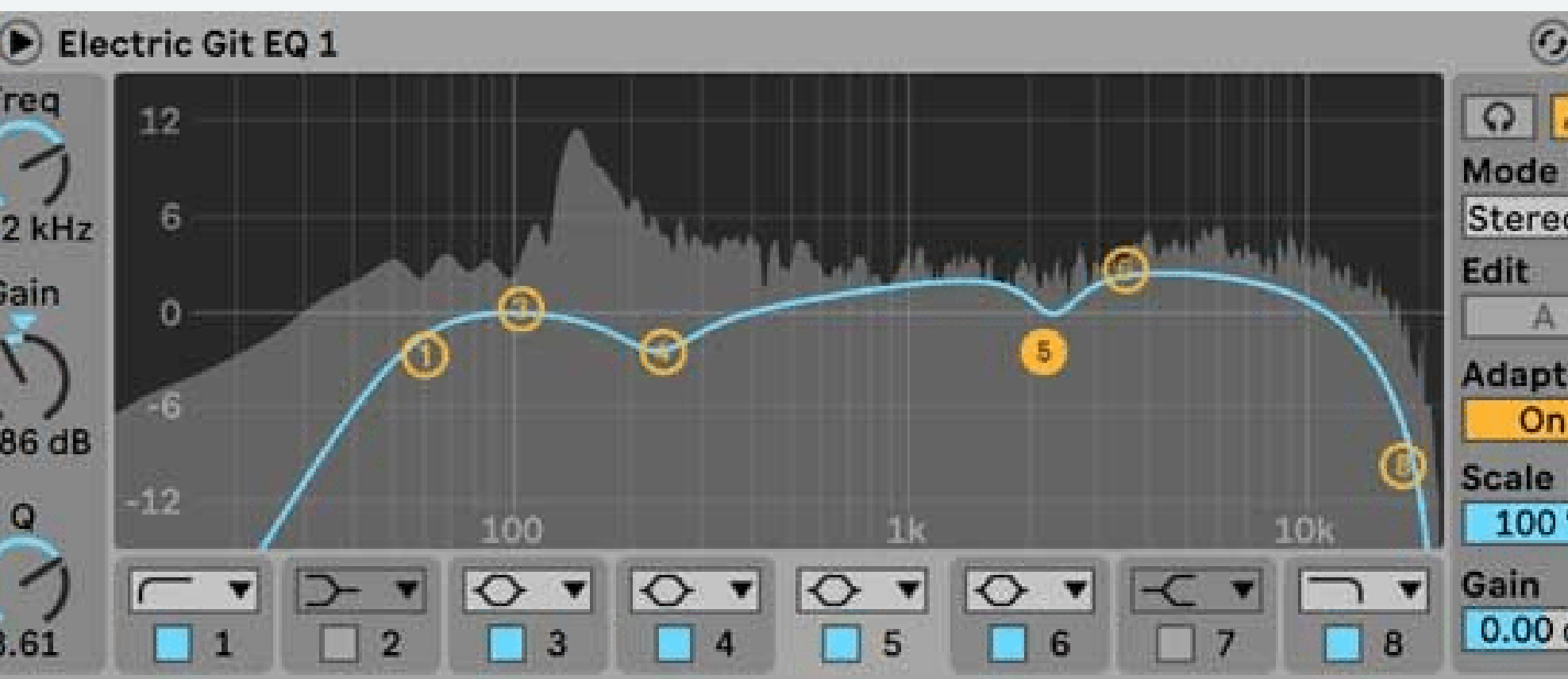


## 2. ENHANCING ELEMENTS

Once we’ve removed the nasty stuff, you can move on to approach #2, which is to enhance pleasing elements.

For this, I prefer to use an analogue modeling EQ, but it’s by no means necessary.

Here are my recommended plugins:



### Stock EQ -

A lot of DAWs now have a stock EQ that models an analogue unit, don’t upgrade for the sake of it!



Slick EQ -

A great free option.



Slate VMR -

You get two awesome equalizers with this versatile plugin.



## **Waves SSL E-Channel -**

A classic plugin that always sounds great.

Okay, back to the 2nd approach...

Let's use the example of applying EQ to a vocal with a 2dB boost at 6 kilohertz.

That's probably because I thought the aggression in the vocal was nice. I sat down and I thought, "What about this vocal do I want to enhance?" It already sounded clear, but I wanted to enhance the upper mids and give it a bit more aggression and treble, which is why I've got a boost here.

So there you go, we enhanced the pleasing elements that are already there.

You can't introduce new elements. That's not how EQ works. You can only enhance what's there, so that's why you need to make sure it's a good recording and you like the tone in the recording phase.

One more thing...

This is also the phase where you are most likely to use shelves.

If something sounds too bright, you could cut a couple of dB's at 10kHz with a high shelf. Or if



something sounds too bassy (but you don't want to completely remove the frequencies with a filter) use a low shelf to reduce everything below 300Hz. But be more careful when using shelves to boost frequencies. I wouldn't recommend using a low shelf to boost bass. But you can certainly use a high shelf to add a bit more air to an acoustic guitar or vocal.

### 3. MAKE THINGS SOUND DIFFERENT

This approach is just to make things sound different. For example, you could filter out all of the top end and low end on a vocal to give it that 'telephone' sound.



Making things sound weird and different with EQ is a great way to add interest and variation to your mix – especially if you only do this in specific sections or phrases.

## 4. CREATE SPACE IN THE MIX

And then finally, we've got Approach 4, which is to create space in the mix using range allocation.

This is a really good way to create separation and space in your mixes. Here's how:

Essentially, this is the act of never boosting two parts at the same frequency.

Instead, once a frequency range has been 'allocated' to a particular part, you probably want to cut that frequency in other instruments.

THIS is how to use EQ to create separation and clarity in your mixes.

By cutting frequencies in some instruments and boosting them in others, you can create space in the mix and give each part its own place to sit, its own pocket in the frequency spectrum.

The following tips are good starting points and recommendations that I've used to build my mixes. As always, experiment with what sounds best for your tracks. And remember – getting a good sounding recording at the beginning with your mics and preamps will go a long way in helping you get a

great sounding mix once you start using these techniques.

## KICK



- Cut everything <25hz to remove unnecessary rumble
- Boost to taste 80 - 200hz & 500-1000hz for punch
- Cut 200 - 450hz to remove muddy frequencies
- Boost 5-10khz for click & cut through the mix



# SNARE



- Cut everything <100hz to remove unnecessary rumble
- Boost 150 - 250 hz for body & 1.5-2.5 Khz for attack
- Cut 400 - 600 hz to remove boxy frequencies
- Boost 5-8 khz for sizzle and clarity

# VOCALS



- High pass >60Hz to remove unnecessary lows and rumble.
- Cut 200Hz slightly with a wide q to reduce muddy frequencies.
- Boost 500Hz to amplify vowels and make them cut through.
- Cut 3-5KHz to reduce the harsh sibilances.
- Use the dynamic EQ mode that turns eq smart and cuts only when the sibilances hit the threshold.
- Boost >8k with high shelf filter to add some air

# PIANO



- Cut below <60Hz to remove rumble/sub & avoid clash with 808/bass
- Cut 150 - 250Hz to cut muddiness & 2-3KHz to make space for vocals/guitar.
- Boost 3k-5k for presence and use high shelf to add clarity and sharpness in the tone

# MID-SIDE EQING



Mid-Side (M/S) EQ is an equalizer process that encodes a stereo signal into separate mono and stereo channels. The mid (mono) channel contains information identical in both the left and right channels. The sum of the left and right channels creates a mono signal ( $L+R=Mid$ ). Conversely, the side (stereo) channel contains information that differs between the left and right channels. The difference between the left and right channels creates a stereo signal ( $L-R=Side$ ). When combined, the mid and side channels produce a full stereo field.

Once encoded into M/S, you can process the mid information independently from the side information. Both channels are then decoded or converted back into conventional left and right stereo channels. This technique allows you to EQ your mix with greater flexibility and precision.

## 5 MID-SIDE EQ TECHNIQUES

The applications for mid-side equalization are countless. You can use it for corrective mixing work, mastering, sound design, creative effects, and more. The creativity is in your hands!

Here are five simple mid-side EQ techniques that will improve your mix. For these mixing tips, we'll use

the FabFilter Pro-Q 3 equalizer. The green filter is the mid band, and the blue filter is the side band.



## 1. ADDING STEREO WIDTH TO CREATE A FULLER SOUND

Use mid-side equalization to create a wider stereo image on a full mix or individual elements. You can create stereo width by changing the balance between mid and side levels. For example, widen a signal by boosting high frequencies in the side channel or attenuating low frequencies in the mid channel.

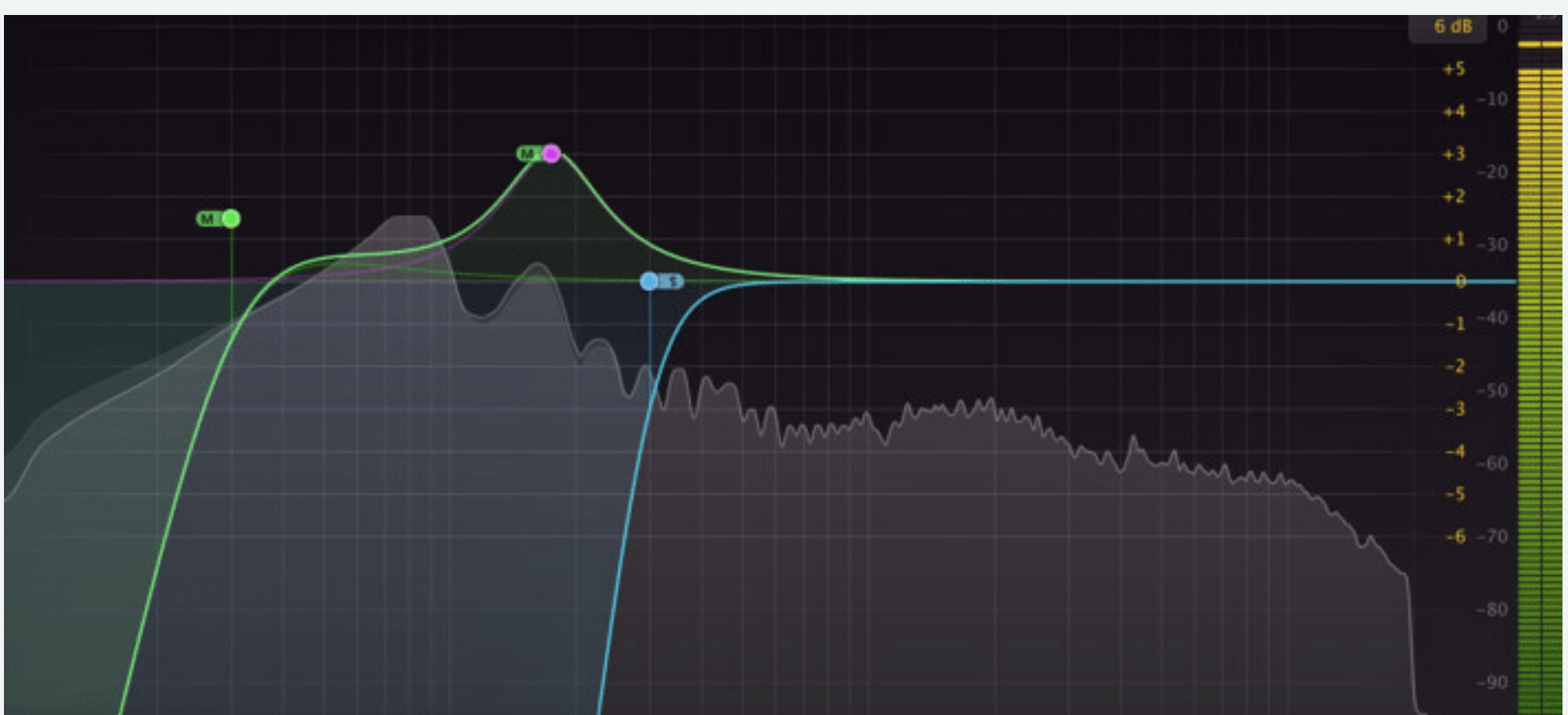
Widening the stereo image will make your mix sound fuller and more enjoyable for the listener. Similarly, stereo widening is also beneficial for sound design. For instance, when layering synths to create a wider, more fuller sound.



The example above widens a synth track by attenuating the mono signal and boosting the stereo signal with a high-shelf filter at 8,000 Hz. This move creates space to layer a mono-focused synth while also creating width. When combined, both layers sound more cohesive, present, and immersive.

This mixing technique may also work better than stereo width plugins sometimes. For example, stereo imaging plugins can affect the phase of a sound, causing it to lose punch. Conversely, mid-side treatment will preserve the impact of the center channel while only widening the side channel.

## 2. MIXING BASS TO ACHIEVE A TIGHTER LOW-END



Modern basslines in electronic music typically have centered low-end and wider harmonic frequencies in the high-end. Keeping low-end sounds like the Kick

and Bass centered maintains their punch and presence.

Achieve a tighter low-end by treating the bass with mid-side equalization. For example, EQ a stereo bassline to preserve a tight and punchy mono signal while retaining the wider, harmonically rich frequencies.

Create this separation by cutting low frequencies from the side channel. The exact frequency cut will vary depending on the content.

A good starting point is anywhere between 80 Hz and 250 Hz. Caution against cutting too much from the sides. It can make the bass sound thin and weak. Use your ears!

The example below removes low frequencies from the side channel at 300 Hz.

It also boosts the mid channel at 175 Hz to give the mono signal more presence in the mix. Pro-Q 3 has a helpful solo button. Use it to hear the area affected by the EQ filter when dialing in the frequency.

This mixing technique ensures the lowest frequencies are mono. It also removes low frequencies from the sides, which causes clarity and phasing issues. As a result, your bass will sound tighter, present, and focused. It also creates headroom, which will give you better results when compressing the overall mix.

### **3. MIXING VOCALS TO SIT PERFECTLY IN THE MIX**

Mid-side processing works excellent at enhancing vocals and getting them to sit in the mix. Try these vocal mixing techniques on the lead topline and backing vocals:

#### **MAKE SPACE FOR THE LEAD VOCAL**

Lead vocals should be front and center in a mix. Keeping lead vocals centered helps them sit in the mix better, gives them more definition, and retains their punch. Instead of boosting the volume of the vocal track, try making space in the mix for the vocal. For example, use mid-side treatment to carve out mid-range frequencies from instruments that conflict with the vocals.

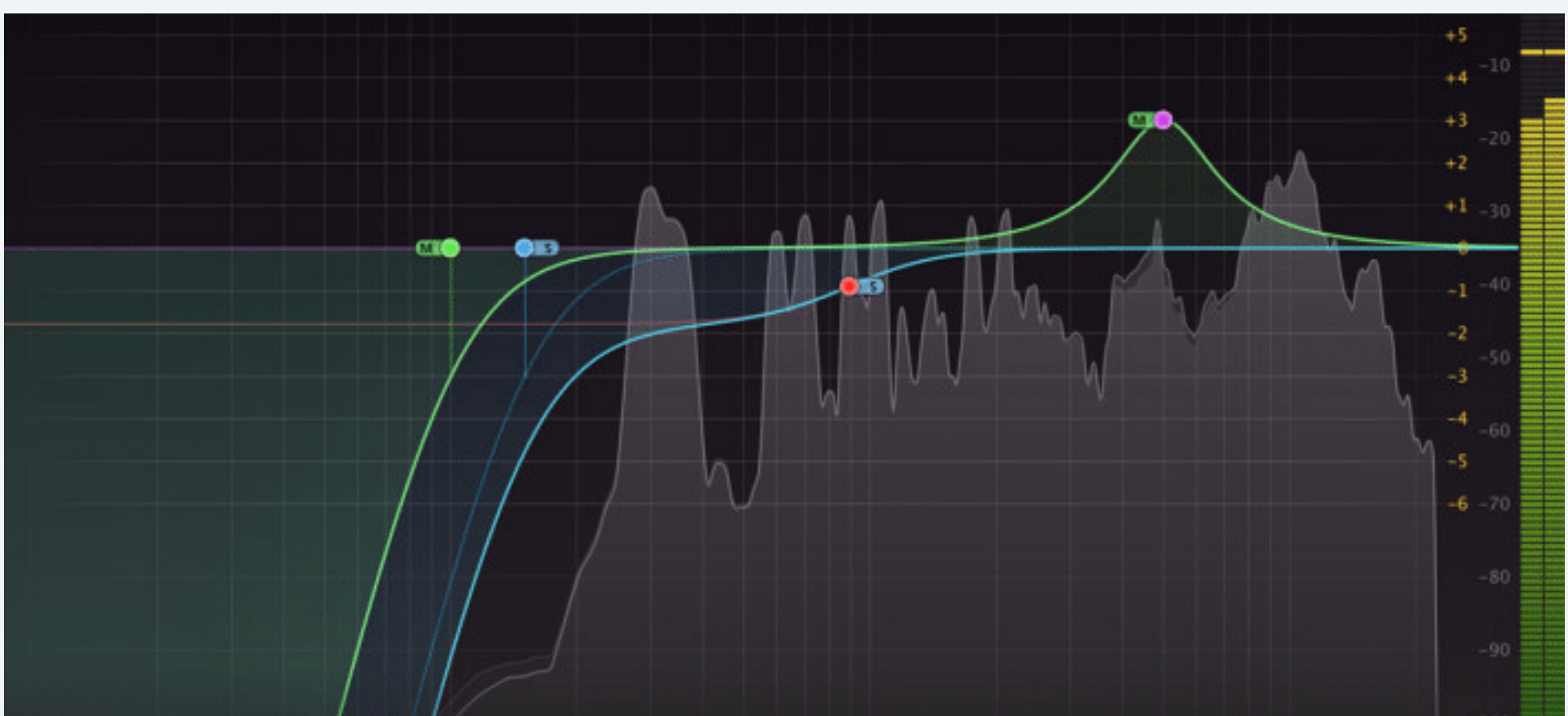
- Start by finding the frequency range where the vocal has the most energy. Find this range by boosting a bell filter on an EQ and then sweeping

across the frequency spectrum. Listen for the dominant frequencies that give the vocal its power. Write the frequency range down and deactivate the EQ.

- Next, group all the instruments with conflicting frequencies in that range.
- Lastly, use a mid-side EQ on the group to cut that frequency range from the mono channel.

These moves create a pocket for the vocals to sit in the mix. Also, dipping the mid frequencies preserves the wider side frequencies, not interfering with the vocals.

## ADD CLARITY AND PRESENCE TO VOCALS



Enhance the lead vocals further by giving them more presence in the mix. The example below uses a mid-side treatment on a vocal topline track. The EQ is boosting the mid-range frequencies on the mid



Enhance the lead vocals further by giving them more presence in the mix. The example below uses a mid-side treatment on a vocal topline track. The EQ is boosting the mid-range frequencies on the mid channel and attenuating low frequencies on the sides. This move keeps the mid-range focused in the center and pushes the high-end information to the sides.

## **BALANCE THE BACKGROUND VOCALS WITH THE LEAD VOCAL**

Mid-side equalization also works excellent on background vocals. Backing vocals are generally panned to the sides to make space for the top line in the center. However, if the backing vocals are conflicting with the lead vocal, use an EQ to make space. Simply attenuate the mono frequencies and boost the sides. Use your ears to balance the topline and the background vocals.



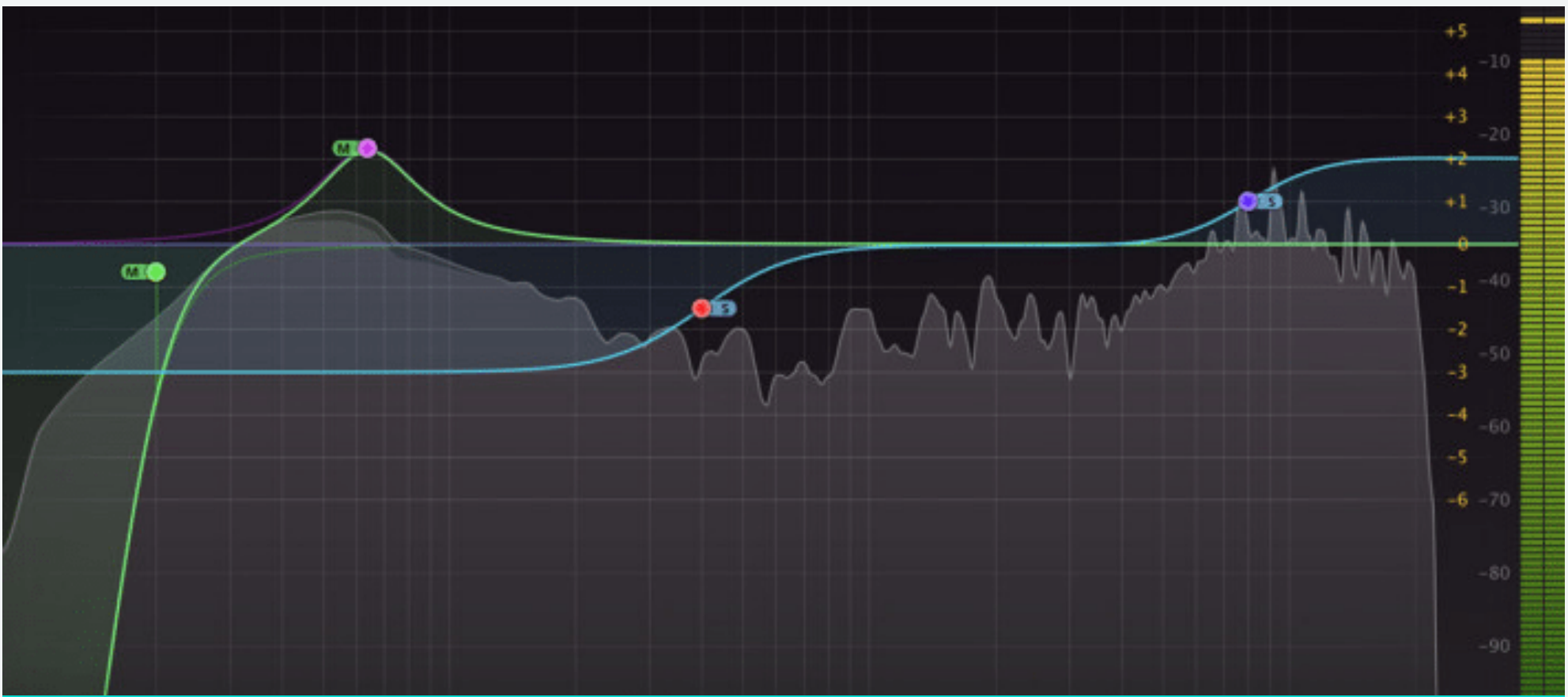
## 4. MIXING DRUMS TO CREATE A TIGHTER, MORE BALANCED SOUND

Mid-side equalization is also effective at tightening and enhancing a drum group. For example, use the mid and side channels to keep your kick and snare centered while adding width to only the percussion. Or maybe you want to lower the volume of high-end frequencies without affecting the kick and snare. There are various scenarios.

Moreover, boosting or attenuating frequencies with a standard EQ will affect both the mono and stereo frequencies. With a mid-side EQ, you have more flexibility over mixing and balancing your drums.

For example, create separation and clarity by cutting or attenuating low frequencies from the side channel. The exact frequency cut will vary depending on the content.

This move gives you the freedom to boost the low-end of the kick and snare to add presence and punch without affecting other drum sounds. You can also boost or attenuate higher drum elements to balance the drum kit and add width.



The example above attenuates the side channel at 400 Hz to keep the kick mono and focused. Also, a high-shelf filter is boosting frequencies at 8,000 Hz to add width. Lastly, a bell filter boosting the mid channel at 65 Hz adds presence and weight to the kick drum. Now, there is more space for the kick to cut through the mix. The overall kit also sounds fuller and balanced.

However, every drum group will be different. Use your ears to make informed decisions on where to cut or boost!

### 3. MASTERING THE FINAL MIX TO CREATE A RADIO READY SONG

Mid-side processing will give your mix more presence, clarity, and balance at the mastering stage. Use an EQ with mid-side capabilities for corrective work, tonal balance, and enhancement.

For example, use the M/S mode on an equalizer to:

**Tighten up the low-end:**

Cut low frequencies from the side channel to clean up the bottom end of the mix. This move also places the kick and bass in mono without affecting the stereo information of other elements.

**Remove problematic resonant frequencies:**

Use the EQ sweep technique to identify unpleasant or nasally sounding frequencies. First, boost a bell curve with a narrow Q and then slowly sweep up and down the frequency spectrum. Listen for harsh-sounding spikes. Next, cut those frequencies by reducing the gain on the mid band. Repeat the process to create a clean and pleasant sound. Also, be cautious not to go overboard. Too many narrow cuts can cause phasing issues.

**Add presence to the mids and highs:**

If the mix sounds muddy, try cutting the low-mids between 250 Hz and 500 Hz on the mid channel. You can also add presence to the mix by boosting the mid channel with a high-shelf filter anywhere above 8,000 Hz.

**Add presence to vocals:**

Boost the mono frequencies of a vocal if it sounds buried in the mix. Again, use the EQ sweep technique to identify the vocalist's strongest frequencies. Then boost those frequencies to add presence. This technique gives you better precision to treat the vocal without affecting the other elements in the mix.

**Improve tonal balance:**

The interaction between all the parts of a song contributes to the overall tonal balance. Correcting tonal imbalances creates a cohesive mix that sounds more natural and pleasing. It also helps the mix translate better on various speaker systems. Balance the tonal spectrum by boosting or attenuating different areas of the mix. Use a broad bell filter on the mid channel and make subtle level changes. Try to achieve perceived equal loudness across the frequency spectrum.

**CONCLUSION**

Mid-side equalization is versatile and effective. You will find many uses for this mixing technique. Remember to make subtle moves and not to go overboard. It's easy to cause more issues when

misused. Also, consider using linear-phase processing when filtering stereo channels to avoid unwanted phase changes. Again, use your ears!

## 10 ESSENTIAL EQ TIPS

Now that you have the strategy down, I want to share some more tips that I have picked up over the years.

You've got the basics down now, but if you want to use EQ like a true professional, keep reading.

### 1. HAVE AN INTENTION

Don't just randomly start boosting and cutting different frequencies to see what works and what sounds good. Instead, decide what you want to achieve first, and then figure out how you can achieve it.

Let me give you an example. I'm mixing a vocal, and it sounds a bit muddy. It's not cutting through the mix enough, and it's kind of clogging up the mix.

Once I know that, I can think "Okay, the low mids are probably where the problem is; that's going to be making it muddy." So I can try to cut at around 400 hertz, and then I can move it around, try 300, try



500, decide on where the best frequency is, where the sweet spot is, and then suddenly the vocal sounds less muddy and the mix opens up. So decide what you want to do first.

## **2. DON'T RELY ON EQ ALONE, ESPECIALLY TO SHAPE THE TONE**

You need to shape the tone in the recording phase, because the tone that you capture there is going to be what you're stuck with. You can use EQ to make it even better, you can use EQ to shape it slightly more, but really the tone is decided in the recording phase.

## **3. PRIORITIZE CUTS, BUT STILL USE BOOSTS**

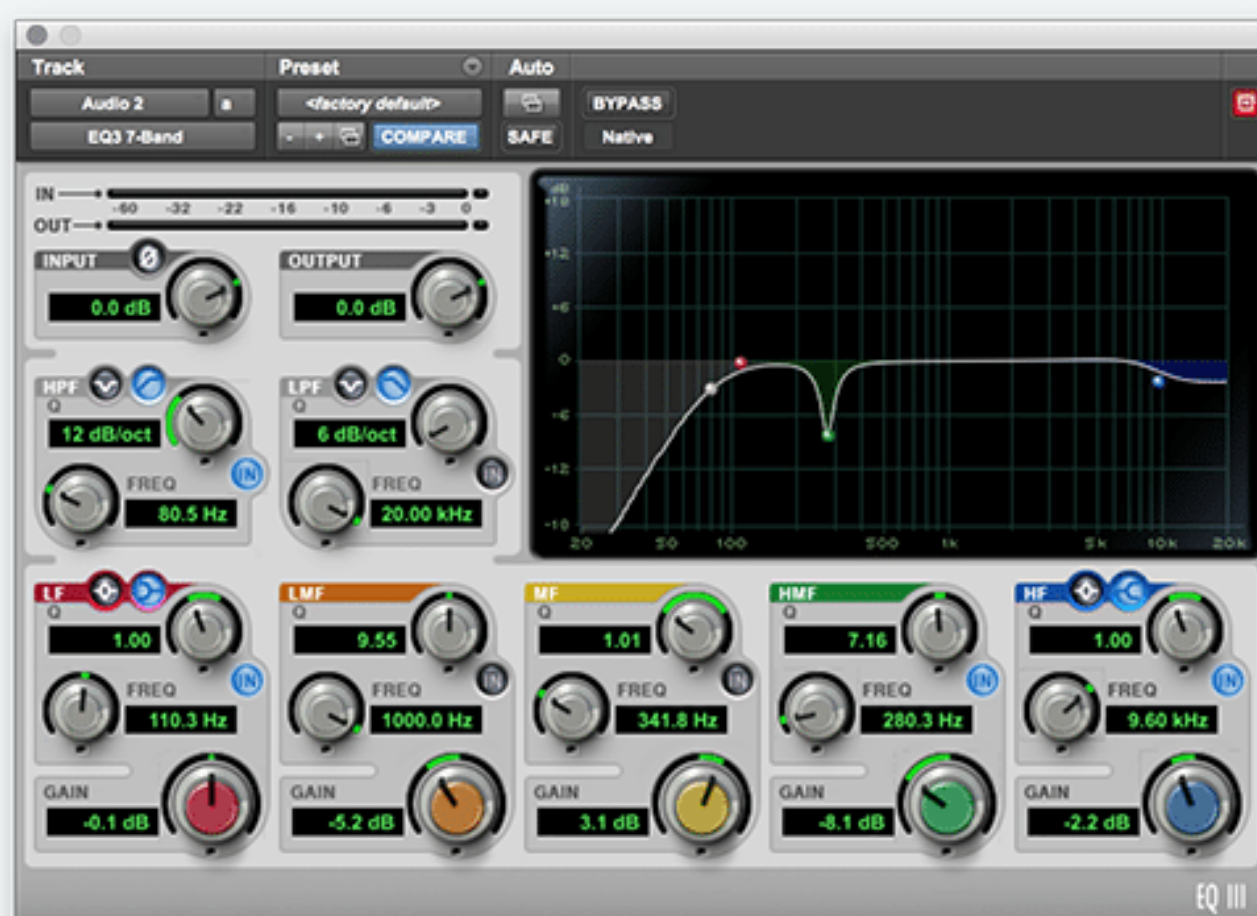
I can remember the first time I learned about subtractive EQ (a long, long time ago).

I understood the principles, but I found the concept extremely daunting.

Whenever you boost with an equalizer, it messes with the phase of your recording and affects additional frequencies. Not just the frequencies that you are boosting.

The idea was that aggressive boosts can quickly ruin your audio, making it unnatural and difficult to mix. Add to this the fact that boosting the volume of a track reduces your headroom, and it's easy to see why boosting should be avoided where possible.

So in principle, we should try to stick to subtractive EQ. Never boost (unless we absolutely have to) and try to only cut.



But in practice, this isn't so easy to stick to... And with modern plugins, the downsides are minimal.

I remember thinking “how am I going to get my vocals to sound more aggressive without boosting on an EQ?”

Listen...

This idea of ONLY using subtractive EQ is ridiculous.

I could never finish a mix in that way. Any time somebody tries to give you a hard and fast rule about mixing, it's probably bullshit.

OF COURSE you can use boosts when you are mixing. If you have an intention, and you need a boost to get there... DO IT.

My advice is to prioritize cuts, but use boosts when you need them.

## **4. AVOID APPLYING EQ IN SOLO**

Instead, try just bringing up the channel a bit.

If you're struggling to hear the changes that you're making, if you're only doing a 1dB or 2dB cut or boost and you're struggling to hear it in the context of the mix, bring that channel up a bit so you can hear it a bit better, because then you'll be able to hear your EQ change, but you're also going to still have the context of the mix.

Whereas if you apply EQ in solo, you can quickly forget about the rest of the mix and how that instrument sits. Plus, it helps to learn how to EQ with the whole mix going.

No one's ever going to hear your mix in solo. They're never going to hear the guitar in solo, so it doesn't

matter if it sounds good in solo. It needs to sound good in the mix.

A lot of the time, things sound really bad in solo, especially with electric guitars. So try to apply EQ in the context of the mix – or you could trick yourself.

## **5. SMALL CHANGES SOON ADD UP**

This one is more for beginners.

When you're starting out, try to stick to cutting or boosting by no more than 5dB. Lots of small changes across your mix soon add up. As you get more confident, you can get more aggressive.

## **6. BE MORE SUBTLE WITH STOCK PARAMETRIC EQS**

Boosting by 10dB or more sounds great when you are using an old analog desk and the EQ section is amazing. It's adding a lovely color to the sound.

A lot of the time with your stock parametric EQ in your DAW, it's going to be different. It's probably going to have some nasty side effects if you start boosting by too much. It's going to start messing with the phase.

So try to be more subtle with stock parametric EQs. If you're using an analog modelling EQ, for example, you can be more aggressive.

## **7. DON'T OBSESS OVER PLUGIN ORDER**

A lot of people ask “Where should I place EQ in my plugin chain? Before or after compression?”

Honestly? It doesn't matter. You are worrying about the wrong things.

Sometimes it sounds better before, sometimes after. Just play around with it – but only for a few seconds.

My go-to is to put surgical EQ before compression, and tonal compression after. It doesn't necessarily sound best – it just suits my workflow.

## **8. YOU CAN'T POLISH A TURD (BUT YOU CAN ROLL IT IN GLITTER)**

Only use EQ to remove nasty frequencies (by cutting) or to change the character of a sound and add interest (by boosting). You CAN'T use EQ to make a poor recording sound good. You can't add stuff that isn't there – only emphasize stuff that's already there.



## 9. CREATE INSTANT CLARITY BY REMOVING MUDDINESS

The most problematic frequency range in most home recordings is 250-500Hz (most instruments are heavy in these frequencies).

Your mix will start to sound ‘muddy’ if there are too many of these frequencies evident.

A gentle, wide cut of 3dB between 250-350Hz on the muddy tracks is a great place to start.

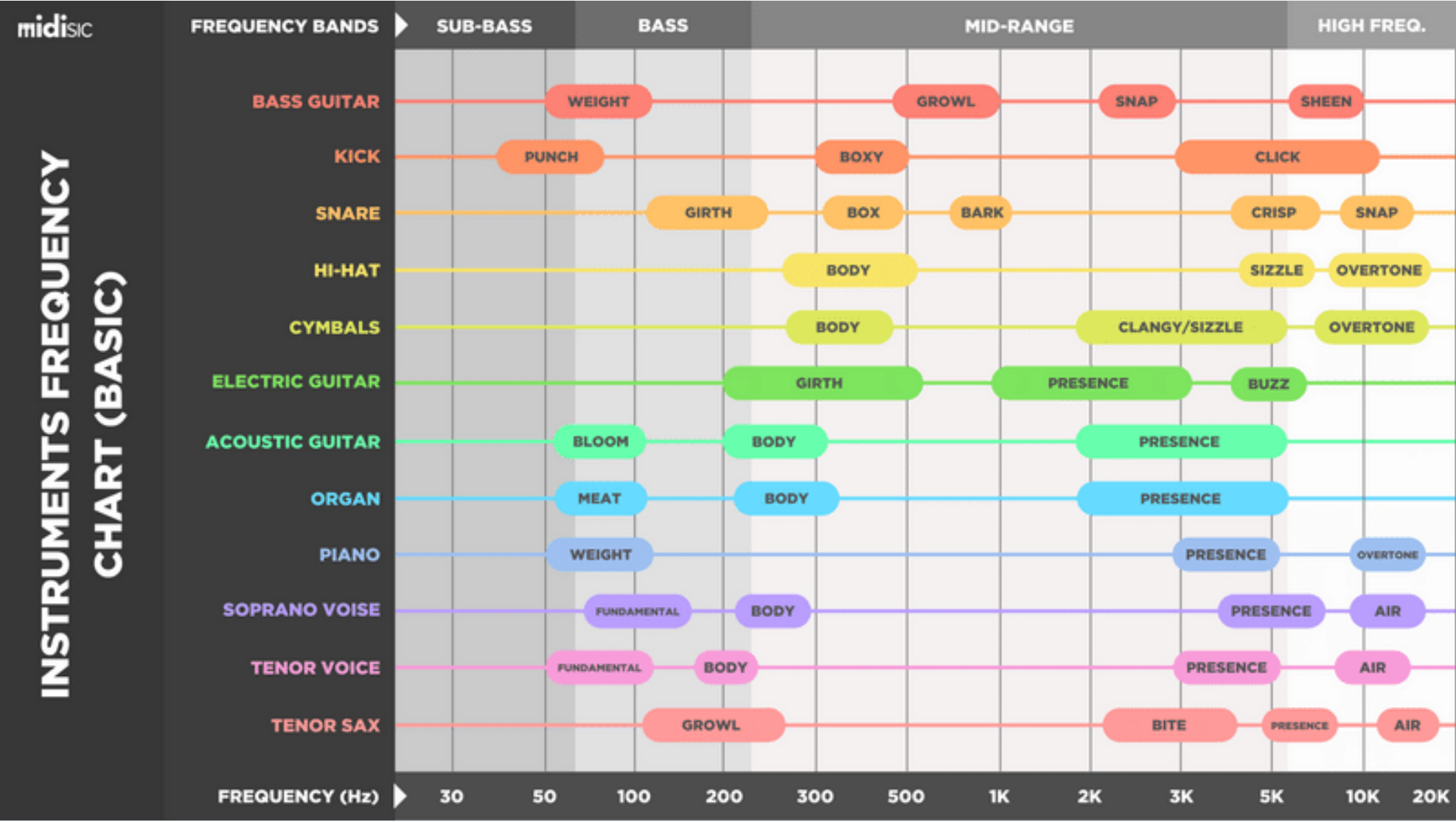
## 10. MIX IN MONO

When applying EQ to your mix, do it in mono. This helps with range allocation and preventing phase issues. It forces you to create space and separation with EQ, instead of relying on panning.

It will make a huge difference, trust me. Start mixing in mono, and when you do eventually start panning tracks towards the end, the space in your mix will be immense.

That’s it for the EQ Tips, now let’s end the EQ Chapter with Extra Tip

# EXTRA TIP (FREQUENCY CHEATSHEETS)



At the end, I want to talk about a couple of frequencies that you should keep an eye out for. If you look at this chart you'll get to know the main frequencies of a lot of different instruments. Let's see how this instrument frequency chart is helpful to us

At the bottom, we have a frequency range that goes up to 20k. So what they're telling you here is for these instruments. Here's where you can find a specific character of these instruments

So for bass guitar, if you really want to make it

sound big and fat boost right around 65-70 hertz or if you make it, sound growly, boost around 700-800 hertz.

Same with every instrument mentioned in the chart like if you want your kick to get really punchy, boost right here around 50 hertz around 300hertz for more of a boxy sound, etc.

So this shows a bunch of different instruments and kind of where their sweet spots are. But remember, it depends on the particular instruments you're using

These charts are great for learning, but make sure to keep them from becoming a crutch. These are great learning tools, but shouldn't be referenced constantly.

Eventually, you want to have memorized this chart by heart. Your workflow will get much faster, and you won't be overusing your EQs (which can take the life out of your music). Also, avoid ANY EQ charts that are specifically about one instrument.

Not all sounds are created equal.

No guitar sounds like other guitars.

No one snare sounds like other snares.

Using an EQ chart that's for just one instrument will give you lots of information, yes, but it's info that doesn't apply to YOUR version of that instrument. Since you're making decisions based on false information, it will make your mixes sound worse.

Okay that's it for the Equalization Chapter, I know it was lengthy, but that's what it should be, so without wasting any time let's go to the next chapter.

## ***NOTE***

As you move through the mixing process and keep adding processors such as EQ and compression, you will inevitably change the initial balance you made with only volume and panning.

That's why it's always a good idea to keep rebalancing the faders as you move through the mix. Even if you keep your gain structure of the plug-ins relatively perfect, you will still need to keep adjusting the volume. Mixing isn't a complete step-by-step process. It's more a set of steps and guidelines you follow while constantly adjusting and reacting to your mix decisions.





**COMPRESSION**

**10**

Compressors reduce the difference between the loudest and quietest parts of the audio its processing. They allow you to control, color and manipulate the dynamics of audio. They're powerful tools but using the wrong setting can suck the punch out of your music. So when should you use a compressor and how should you tweak the parameters to work with the material?

The first step is to decide whether the compressor is the right tool for the job. Using a compressor to balance out audio with a large dynamic variation will give a very uneven sound. The softer parts will sound unchanged and the louder parts will be squashed in an unnatural way. Automating a gain plugin is a much more natural approach to level out audio with large dynamic variations. Once your channel has an even volume balance, you can use a compressor to thicken the sound and control the transients taking an artistic approach rather than a corrective approach.

Any compression you use should preserve the character of your audio transients. Even when you're using a charismatic compressor with lots of color or applying intense settings for obvious effect, your

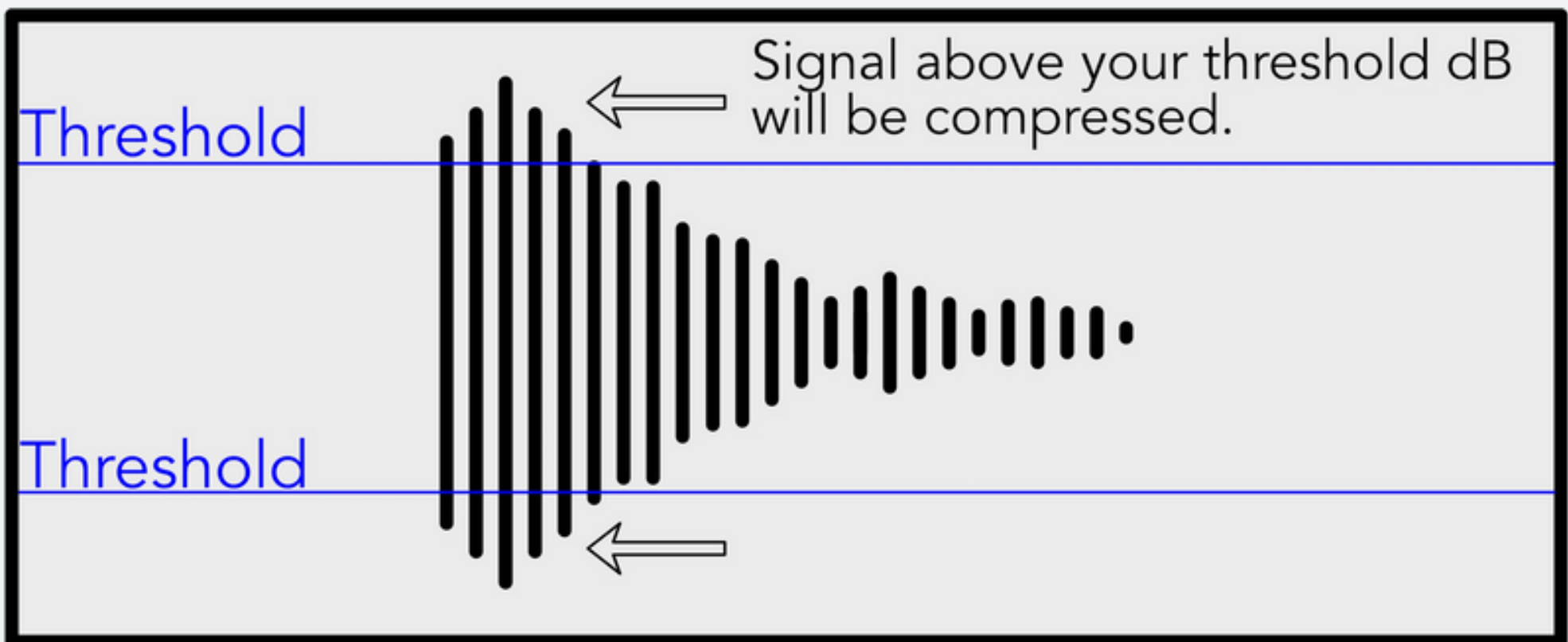
goal should be to musically bring out the natural dynamics of your signal. As you’re tweaking your compression settings, think about what you’re trying to achieve. Are you adding character or adjusting the transients and dynamics range? How has the compression changed the relationship between the loud and soft parts of the audio? How have the transients changed? Do these settings slam the audio in too much of an obvious way?

Think about the vibe and purpose of what you’re trying to achieve. The table below will give you an idea of which compressor is the right tool for the job.

Vibe and Purpose	Compressor Characteristics	Which Compressor?
Punchier Transients	FET circuitry, analogue emulation.	Universal Audio 1176
Dynamic Control	Clean sound, no colour.	Fabfilter Pro C2
Parallel Compression	Opto circuitry. Works slower adds a nice pumping effect thats beefs up the sound.	Universal Audio LA2A
Add Colour & Harmonics	Analogue emulation.	Fairchild

# COMPRESSOR PARAMETERS

## 1. THRESHOLD



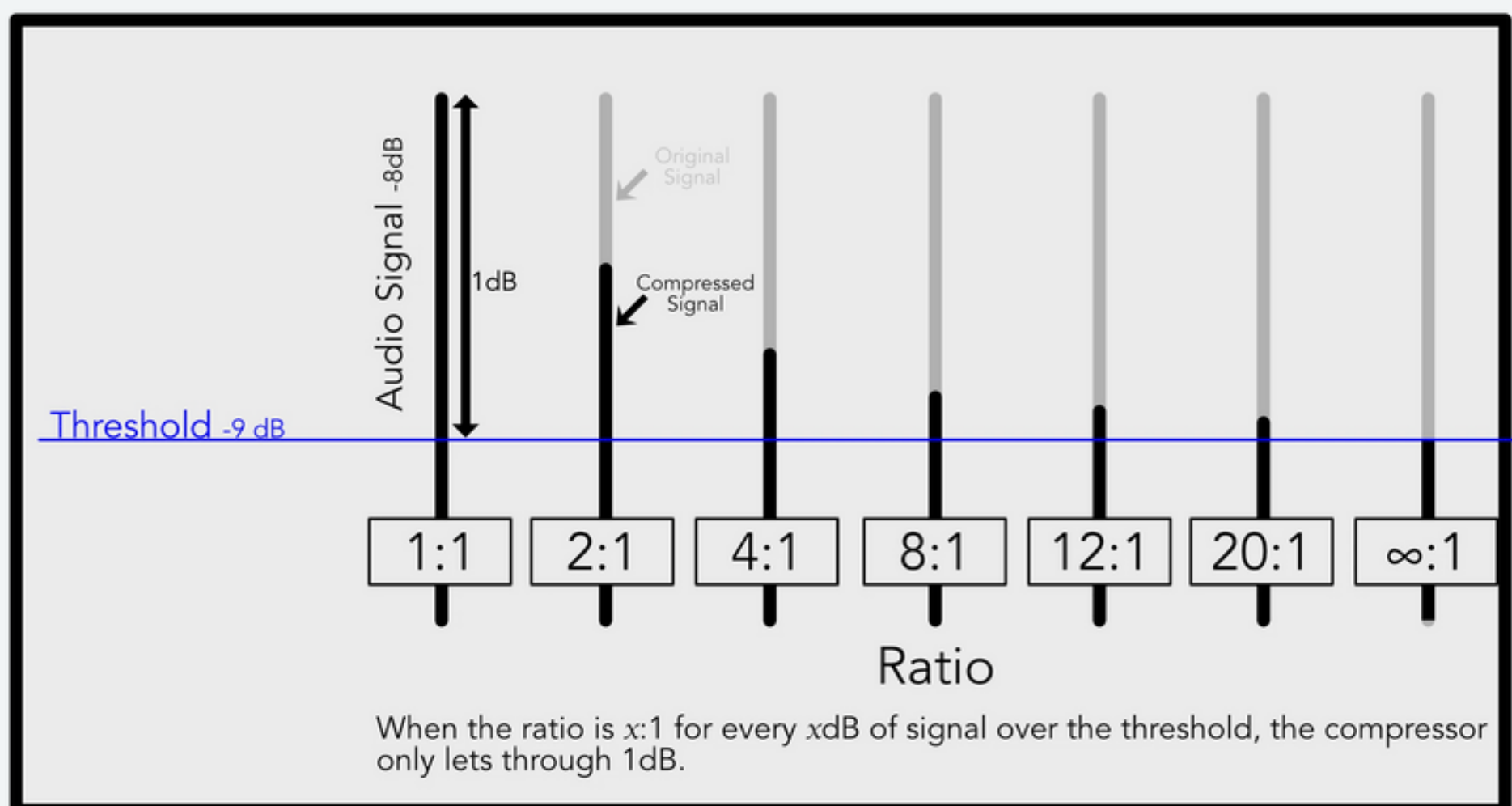
What does threshold mean in a compressor?

The threshold sets the level at which the compressor will start reacting to the audio signal. If your audio is peaking at -10dB and the compressor threshold is set to -4dB, the compressor simply won't react to your audio at all. If you wanted your compressor to only look at the peaks of your audio (-10dB again) then you would set the threshold to around -13dB. If you wanted to heavily compress your audio then you would set the threshold to react to both the loudest and quietest parts of your audio.

## 2. RATIO

Ratio sets the amount of gain reduction. The higher the ratio, the more extreme the compression. To give you a ballpark idea, 1:1 is no compression, 2:1 is

3:1 is moderate, 4:1 will be substantial, 8:1 will be very noticeable, and anything above 10:1 can sound rather slammed unless used in tandem with a very light threshold.  $\infty$ :1 is limiting and nothing goes over the set volume.

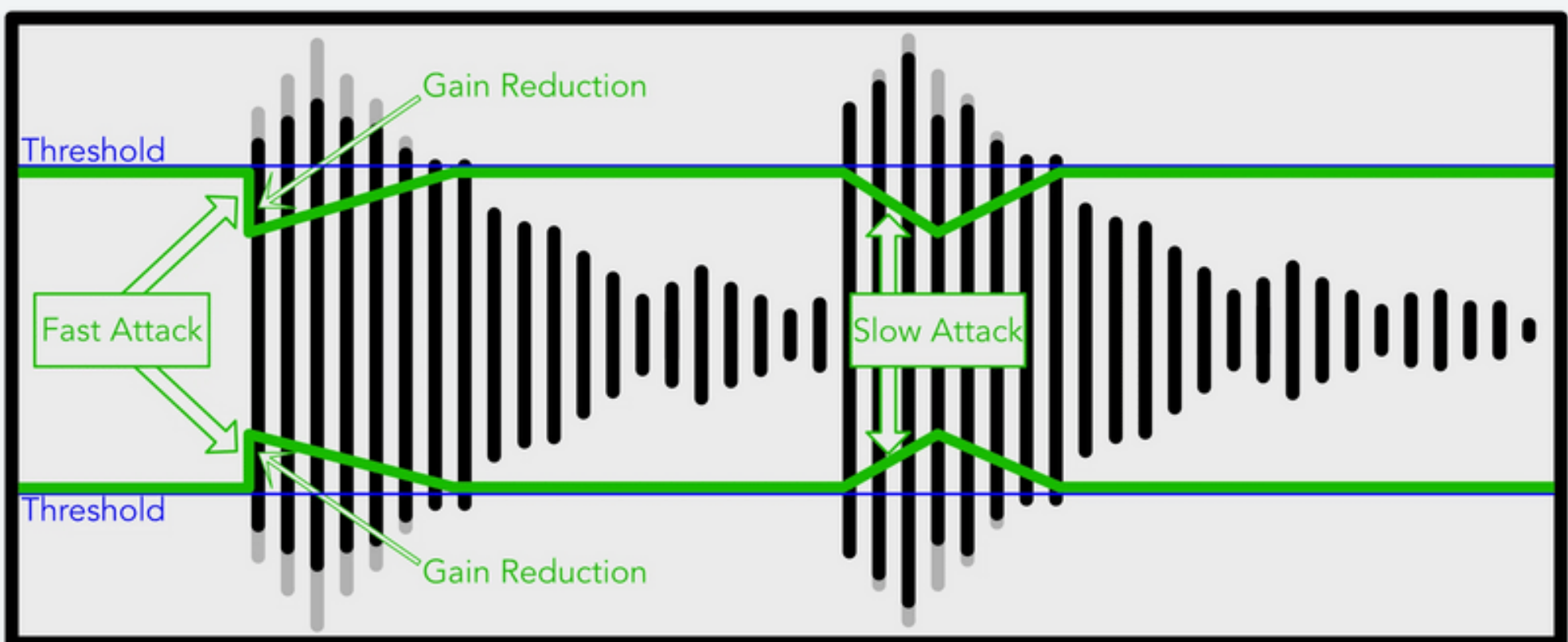


### 3. ATTACK

Attack is the amount of time it takes the compressor to react to the incoming signal. If the attack is immediate or super fast, the compressor will catch the transients of your audio material. This can be really useful when the transients sound a bit sharp. You can use a compressor to make the transient feel a bit blunter, and therefore a bit thicker.

If you want to add some leveling to your audio but you want to leave the sharpness of the transients intact, go for a slower attack time



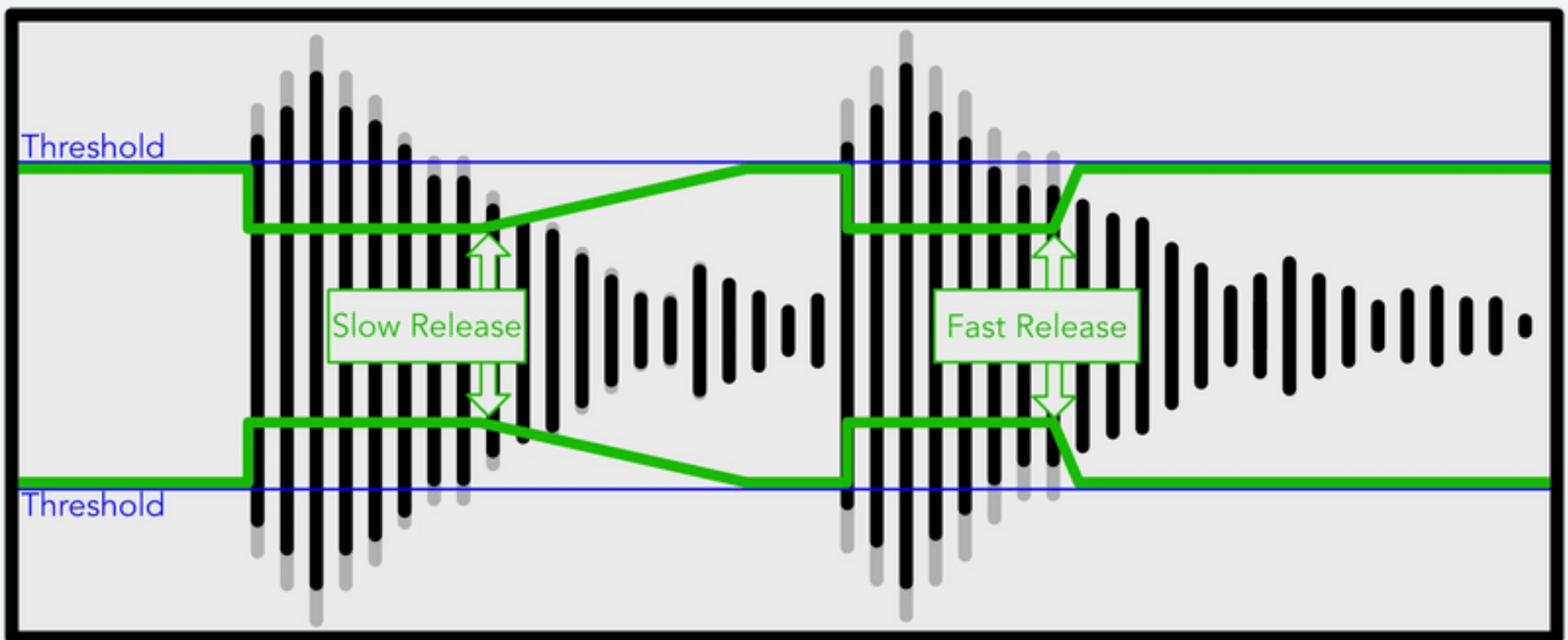


## 4. RELEASE

What does release mean in a compressor?

Release is the amount of time it takes for the compressor to return to a non-compressing state. A fast release will mean the compressor will stop compressing quickly after the audio is no longer over the threshold. A slow release will mean it takes longer for the compressor to stop working once the audio is no longer surpassing the threshold.

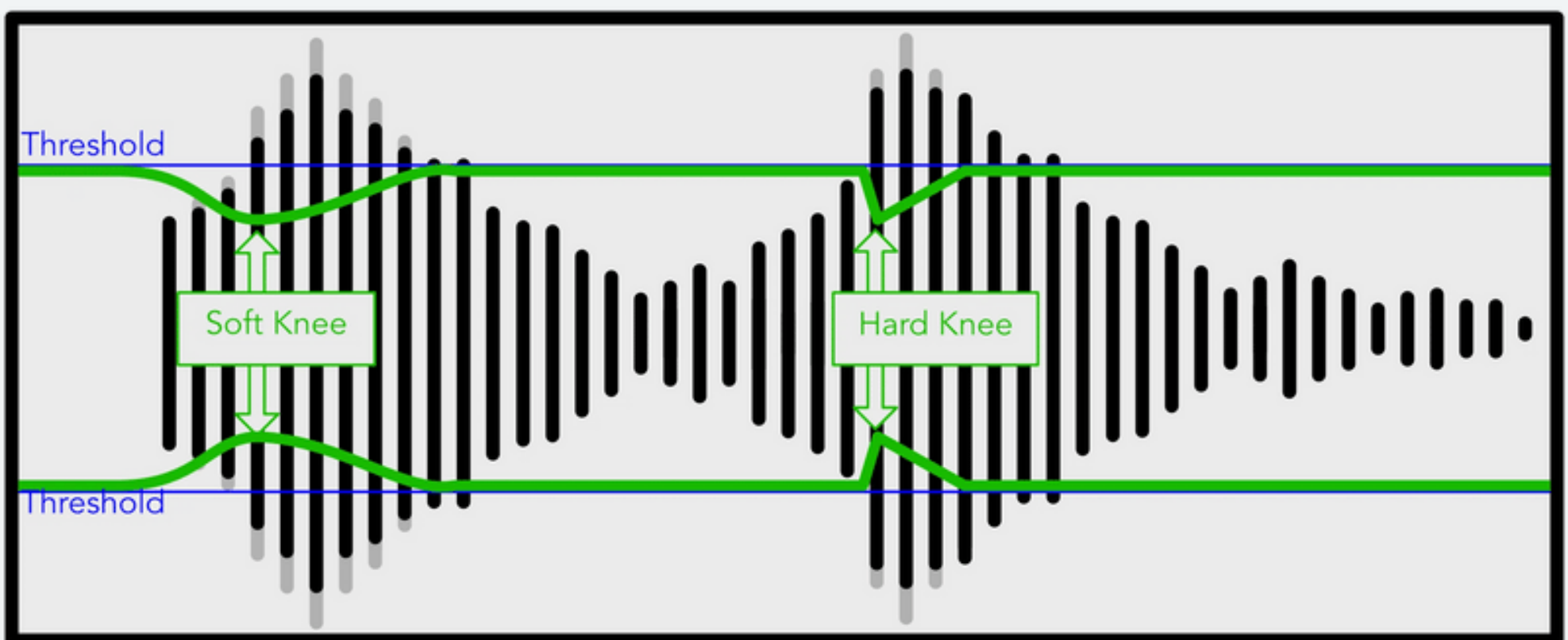
I like to set the release to work rhythmically with the audio that's being fed into the compressor. For example, If I'm compressing a kick, I will set the release to just long enough that the compressor returns to a neutral state before the audio re-triggers the compressor again. For a more obvious and pumping sound, you might choose to go for a longer release.



## 5. KNEE

Soft knee means the compression will be applied gradually as the signal approaches the threshold.

Hard knee means the compression will be applied quickly as soon as the audio surpasses the threshold.



## 6. OUTPUT GAIN

The compressor is effectively turning down the louder parts of the audio and leaving the quieter parts untouched, thus reducing the difference between the louder and quieter signals.

So this will turn down the overall volume of that channel. So you'll most likely want to use the output or makeup gain to bring the level back up so it works well in the mix. I like to bypass the plugin on and off to make sure the channel is still well balanced in the mix.

Type Of Sound	Ratio	Attack	Release
Upfront Vocal	3:1 to 4:1	15 to 30 ms	40 to 80 ms
Agressive Bass	3:1 to 4:1	16 to 20 ms	50 to 300 ms
Tight Drums	2:1 to 4:1	1.5 to 6 ms	50 to 250 ms
Piano Weight	3:1 to 8:1	20 to 75 ms	0 to 30 ms
Controlled Guitar	3:1 to 4:1	10 to 20 ms	40 to 100 ms
Thicker Sound	2:1 to 4:1	3 to 12 ms	20 to 75 ms
Parallel Compression	8:1 to ∞:1	0.5 to 1.5 ms	50 to 150 ms

Fast attack and release times can sometimes cause low frequencies to distort. Low frequencies have long waves so the fast attack causes the gain reduction to begin working within one cycle of the wave causing it to clip. The problem is, sometimes you need a fast attack time to attenuate the transient of the audio.

My preferred fix for this is to increase the 'lookahead' time. This means the compressor begins the gain reduction a few milliseconds before the transient, allowing the low frequencies to pass through undistorted.

# TYPES OF COMPRESSORS & THEIR CHARACTERISTICS

## 1. FET COMPRESSION

### WHAT IS FET COMPRESSION?

FET (or Field Effect Transistor) compression came about when audio units started to replace large tubes with smaller transistors.

### WHAT IS FET COMPRESSION GOOD FOR?

As a general rule, FET compressor's slowest attack time tends to be faster than other compressors such as a Variable Mu. FETs themselves give a really punchy tone and sound great across the board whether it's drums, vocals, or guitars you're recording so it's no surprise to see at least one in most pro studios!

Their sound tends to be rich with a lush distortion to it when driving the compressor. The Universal Audio 1176 is arguably the most famous FET compressor and can be bought both as a rack unit and plug-in form.

### BEST FET COMPRESSOR PLUGINS (OUR PICKS)

## **1. SOFTUBE FET COMPRESSOR**

The Softube FET Compressor plug-in is a digital model of a classic FET design. It gives a warm sound whilst keeping vocals and drums punchy. Its intuitive controls mean it's also great for sidechaining and parallel compression.

## **2. ANALOGUE OBSESSION FETISH**

Analogue Obsession's Fetish is a FET style compressor, with a super fast attack time. Its design means it's really easy to use if you're just starting out. It's also compatible up to a sample rate of 192khz.

## **1. VCA COMPRESSION**

### **WHAT IS VCA COMPRESSION?**

A 'Voltage Controlled Amplifier' tends to offer the most aggressive gain reduction, which makes it a good choice if you're trying to keep in check any loud transients or spikes in volume.

Technically a VCA isn't actually an amplifier, rather it uses an attenuator to reduce the volume when it's fed a certain level of voltage.

### **WHAT'S VCA COMPRESSION GOOD FOR?**



VCAs are generally suited to your workflow if you're trying to tame any intense peaks in your track due to its snappy response, however, it won't be as suited to averaging out the level of a song as a Variable-Mu or Optical.

The SSL G Series is one of the more famous versions of a VCA compressor. This can be bought in outboard form or a more cost-effective plug-in version SSL G channel strip by Waves.

## **BEST VCA COMPRESSOR PLUGINS (OUR PICKS)**

### **1. SSL G CHANNEL STRIP**

The SSL G Channel Strip is modeled on the classic Solid State Logic SL 4000 console. The G Channel Strip can take a little while to get used to but the results it produces are worth the time getting to grips with it. It also includes EQ controls, try using some of their presets to start with and tweak from there.

### **2. ACUSTICA TAN**

The Acustica TAN is a free VCA style compressor with an easy-to-use interface. This model includes a 'ShMod' (Shape Modulation) function which allows

you to tweak the attack curve giving you an added level of customization to the compressor's behavior.

### **3. OPTICAL COMPRESSION**

#### **WHAT IS OPTICAL COMPRESSION?**

An Optical Compressor (or Opto-Compressor) uses a light source to control how much gain reduction is applied.

The speed of light is extremely fast, but an Opto compressor is actually quite slow and smooth.

Depending on the level of the signal, an internal light bulb will start to glow getting lighter with the more signal it receives. Because of this, an Opto compressor works off the average signal being received.

#### **WHAT'S OPTICAL COMPRESSION GOOD FOR?**

You wouldn't get a great result trying to control short, sharp transients in the same way you would with a VCA for example, but you would be able to create a smooth compression at much higher ratios. The smoother reaction creates a much more pleasant sound as the signal won't be dramatically squashed.

If you've ever seen a 'Peak' and 'RMS' options in your DAW, think of an Opto as a compressor that reacts best to an RMS signal.

Your RMS display essentially averages out the signal coming through, whereas Peak will display, well, your peaks. In this way, it's easy to visualize how an Opto works.

Opto compressors can really add an element of coloring to your sound as well. Check out the Joe Meek range of Opto compressors. Meek himself was a pioneer of using compression as an 'effect', and aside from having a very colorful history himself, produced some great pieces of gear.

## **BEST OPTICAL COMPRESSORS (OUR PICKS)**

### **1. WAVES RENAISSANCE COMPRESSOR**

The Waves Renaissance Compressor is a smooth all-rounder. It has really easy-to-use Vintage Opto and Electro compression modes which give an added warmth to your vocals or instruments.

### **2. BLOCKFISH (INCL OPTO+VCA MODES)**

The Blockfish Channel Compressor comes with an

Opto and VCA function, which is great if you are just starting out and want to A/B the two to hear the difference. Its design is easy to understand and once you're used to it you can take the front plate and make your own adjustments to its circuit board.

## **4. TUBE (MU VARIABLE) COMPRESSOR**

### **WHAT IS TUBE COMPRESSION?**

A variable Mu (or tube) compressor is one of the oldest types of compressor.

A simple way of thinking about a Variable Mu is that the louder the signal, the more compression the unit will add. This type of compression relies on the tubes themselves to control the gain reduction and therefore the process tends to be quite smooth.

### **WHAT'S TUBE COMPRESSION GOOD FOR?**

Because of the less aggressive characterizing of a Mu, it is suited to situations where you need to equal out your source rather than tame your loud transients (in which case a VCA may be more suitable).

These types of compressors can add a bit of color

to your mix but also help level out thinner sounding instruments as they add a certain amount of warmth which is suited to instruments such as guitars or drum overheads.

The slower attack time on a Variable Mu can really help ‘glue’ your track together which can create a much more balanced and smooth mix.

## **BEST TUBE COMPRESSORS (OUR PICKS)**

### **1. SONIMUS TUCO COMPRESSOR**

The Sonimus TuCo Compressor is another great all-rounder and is suitable for both mixing and mastering. Much like an analog Variable Mu, you’ll notice subtle reductions at low levels with a punchy tone when you crank up and start to drive the compressor.

### **2. KLANGHELM MJUC JR**

The Klanghelm MJUC JR is a great first step into compression plug-ins. Klanghelm themselves have a really good reputation and the MJUC JR showcases why. Great for intense pumping effects but also with the ability for subtle and gentle compression, this is a great addition to any engineer’s collection of plug-ins!



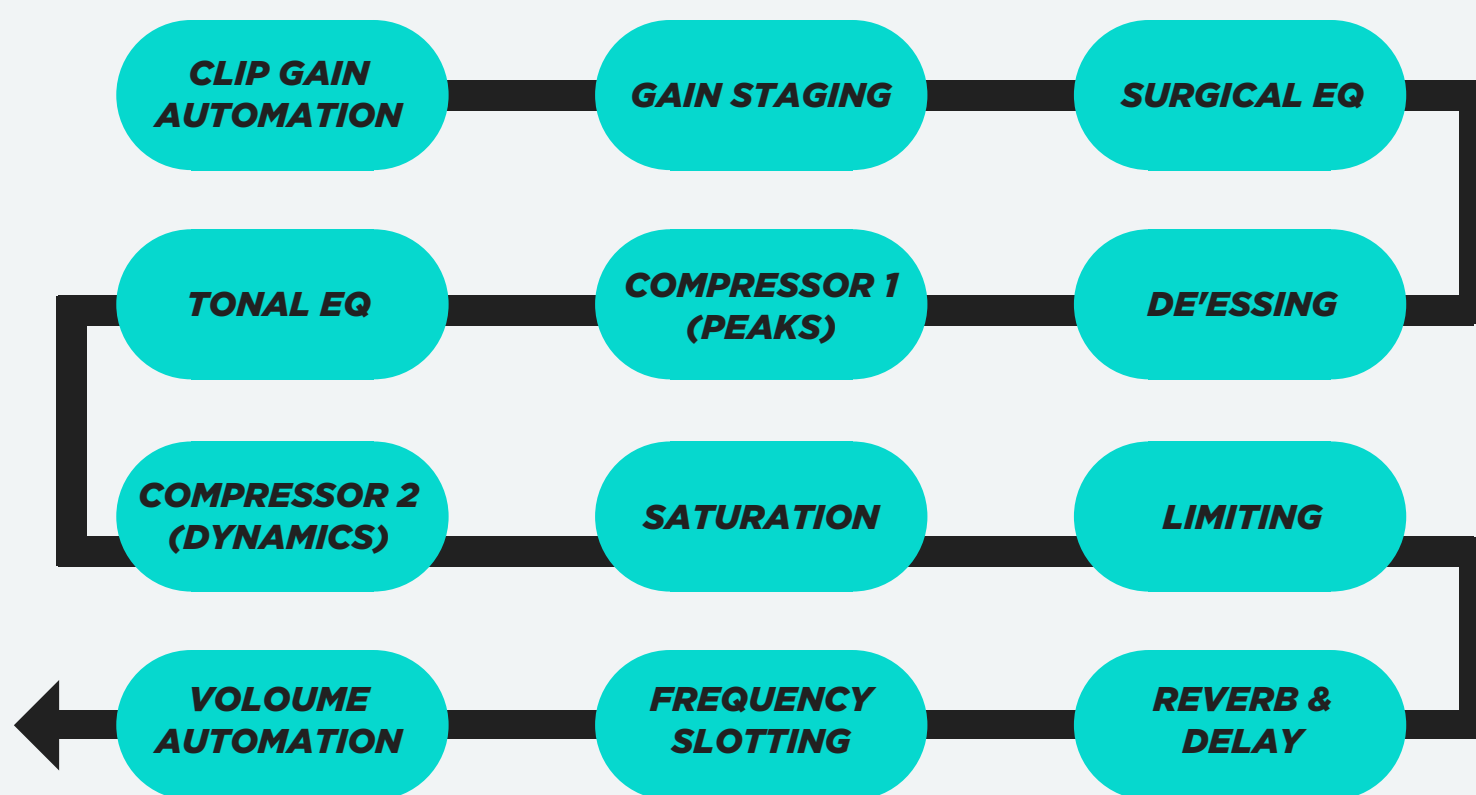
## COMPRESSION IN SERIES

To achieve subtle, natural-sounding compression on your vocal it's best to use multiple compressors throughout the mixing process (this is called 'serial compression').

Rather than using one heavy compressor with a 10:1 ratio that's applying 10+ dB of gain reduction, it's better to use multiple compressors that each chip away at the vocal applying 2-3dB of gain reduction.

For some genres, only one compressor is needed (Jazz, some Rock, Acoustic, etc). But when heavy processing is necessary (Pop, Hip Hop, R'n'B, Electronic, etc) it's better to apply compression in stages.

Here is an overview of my usual mixing system for mainstream, modern vocals:



As you can see, I use compression before AND after EQ.

The first compressor uses ‘tonal compression’ to shape the tone, lightly tame the dynamics and add or remove aggression from the vocal.

The second compressor uses ‘dynamic compression’ to catch the loudest peaks and add more dynamic consistency. We can also shape the tone here too.

## TONAL COMPRESSION

This lighter approach to compression is about shaping the tone of the vocal and adding a small amount of dynamic control just to smooth out the levels.

This approach should use quite a slow attack and release time. You don’t want to compress the

transients too much, as this will push the vocals further back in the mix.

Adjust the threshold so that the compressor is engaging on almost every word, but then use a low ratio to make sure you are only applying 2-3dB of gain reduction.

This is how to compress vocals using a lighter, more musical approach:

- First of all, load up a compressor. Any will do.
- Next, lower the threshold and raise the ratio to extreme settings. This allows you to clearly hear the compressor working.
- Start with a medium attack time of around 15ms and adjust to taste. A fast attack (5ms) will make your vocals sound thick and heavy. A slow attack (30ms) will make your vocals sound punchy and aggressive.
- Dial-in a medium release time of 40ms and adjust from there. Try to get the compressor pumping in time with the music.
- Once you've settled on an attack and release time, bring the ratio down to somewhere around 1.5:1 and the threshold back up to around -24dB.
- Finally, adjust the threshold and ratio until you

are averaging 2-3dB of gain reduction (or higher for heavier music).

The main parameter that dictates the tone of the vocal is the attack time. Spend plenty of time tuning that in, and adjust it in the mix after you have set the compressor.

You could also try using a fast release time, or even setting it to as fast as it can go. This makes the vocals sound loud and aggressive.

Most of the time, though, it's best to use a slower release time and adjust the timing until the compressor seems to breathe with the tempo of the track.

Remember, this approach to compression is about shaping the tone of the vocal, so try a few different compressors and compressor types if you have them.

Spend plenty of time adjusting the attack and release time until you are happy with the vocal. Use a fast attack time for thick, heavy vocals and a slow attack time for punchy, aggressive vocals.

I also recommend using a soft knee compressor for this approach if possible (just turn up the knee parameter if your compressor has one). This sounds more natural and musical. You don't need a hard knee for dynamic control here.



## HOW TO GET LIGHT TONAL COMPRESSION

Here are my go-to compression settings for vocals:

- Ratio: 1.5:1
- Attack Time: 15ms (but up to 30ms for more punch)
- Release Time: 40ms
- Threshold: -24dB
- Gain Reduction: 2-3dB
- Knee: Soft
- Makeup Gain: 2dB

Remember that these settings are only a starting point. A lot of tweaking will be needed, and



sometimes you may have to use completely different settings. But this is what I start with 80% of the time

## **B. DYNAMIC COMPRESSION**

This approach to compression is about catching the louder peaks and reducing their volume. For lighter genres, this type of compression might not be appropriate.

To do this, a faster attack time and higher ratio are needed.

As we are only trying to catch the louder peaks, a higher threshold is also needed. You don't want to compress every word (like with tonal compression), you only want to compress the louder peaks. You still need to be careful not to compress the transients too much, as this will put the vocals further back in the mix. Use a faster attack time, but try not to go below 1ms.

Adjust the release time so that the compressor breathes with the music, or use a fast release time for more loudness and aggression.

This time, the process is slightly different:

- Load up a compressor (any will do).
- Set the ratio to 10:1 (and the knee to 'hard').
- Adjust the threshold until the compressor is only engaging on the louder peaks, not every word.
- Start with a medium-fast attack time of around 3-10ms. You can try using a slower attack time for more aggression, or a faster attack time for more weight (but don't go too far below 2ms as you will put the vocal further back in the mix).
- Set the release time to 20ms and adjust until the compressor is breathing in time with the music. Try using a fast attack time for more aggression.
- Lower the ratio to somewhere around 2:1 until the compressor is applying 2-3dB of gain reduction (or 6-10dB for heavier genres)

## HOW TO GET GREAT DYNAMIC COMPRESSION



- Ratio: 2:1
- Attack Time: 5ms (medium-fast)
- Release Time: 20ms (medium)
- Threshold: -24dB
- Gain Reduction: 2-3dB
- Knee: Hard
- Makeup Gain: 1dB

## COMBINING TONAL AND DYNAMIC COMPRESSION

These two approaches work best when combined. As I mentioned earlier, compression sounds more musical and natural when applied in several stages (serial compression).

For lighter genres where little compression is needed, tonal compression is more appropriate. But for other genres where heavy processing is needed, apply both tonal compression and dynamic compression with two different plugins (or hardware units).

I rarely use dynamic compression on its own for vocals (as it's normally used in combination with tonal compression), but I often use tonal compression on its own for lighter genres.

Here is my go-to plugin chain for a mainstream vocal track:

- Light tonal compression when tracking
- Surgical EQ
- Tonal compression (slower attack, lower ratio, lower threshold)
- Tonal EQ
- Dynamic compression (faster attack, higher ratio, higher threshold)

You can also experiment with using the faster dynamic compressor BEFORE the slower tonal compressor. This means that your slower compressor isn't thrown off by any loud peaks, as the fast compressor will catch them first.

NOTE: Always experiment with plugin order. Depending on what you are trying to achieve, applying plugins in a different order can make a big difference to the sound.

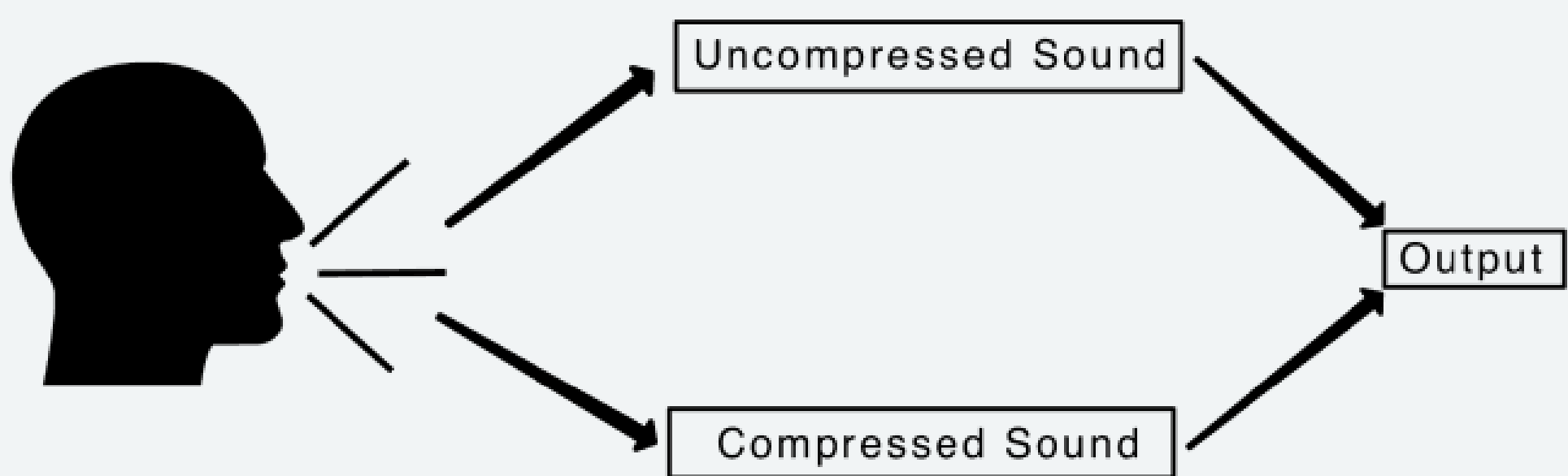
## **WHAT IS PARALLEL COMPRESSION?**

The keyword here is parallel.

How do you usually compress an instrument? Slap a compressor on it, right?

The old-school way of compression is great, but sometimes you don't want the whole thing to sound compressed.

With parallel compression, you duplicate a sound and only compress the duplicated version. So you end up with two different channels. Both of them are playing the same sound, but one's compressed and the other isn't.



## WHY IS IT SO POPULAR?

Tons of mixing engineers love parallel compression. And for good reason!

With parallel compression, you can easily control the dynamics of an instrument without making it sound like it's been compressed. In other words, it's great at natural compression.



It's also really helpful if you want to make an instrument punch through a mix. If you crush heavy parallel compression and mix it in lightly, then you can really highlight a track without making it too loud.

Whether you want natural-sounding compression or obvious, energetic pumping, parallel compression is an incredible mixing tool.

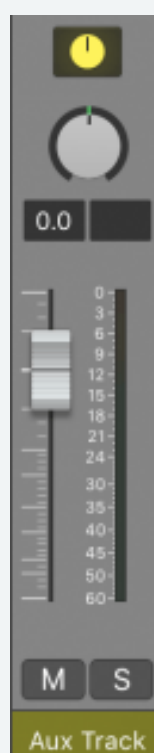
## HOW TO USE PARALLEL COMPRESSION?

First things first, you'll need a track that you want to compress.



This kick drum sounds a little wimpy, so I'll beef it up with some parallel compression.

Then, you'll want to create an "aux channel" in your DAW.



In other words, aux tracks play an exact copy of other channels in your DAW.

Next, go to the audio track you're trying to duplicate. Add a send.



Make sure your send's output is set as your new aux track.

If you aren't sure how aux tracks work, google your DAW's name followed by "create aux track." That should help you find what you need!

Turn up the send's volume and you should have two different tracks playing the same sound!



Next, add a compressor. This is where you'll actually set the compressor and make sure it sounds nice.

## MY TOP 10 TIPS FOR AUDIO COMPRESSION

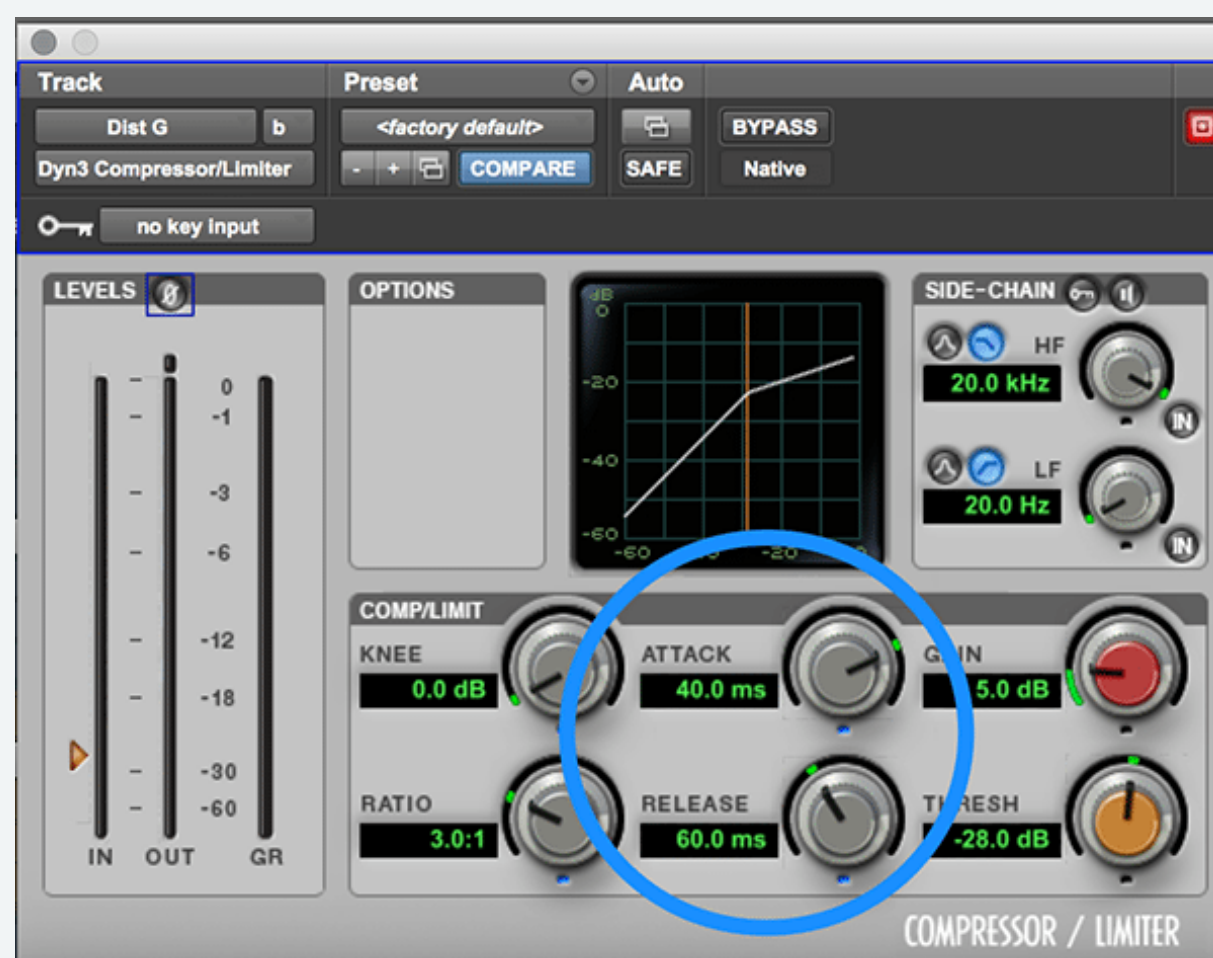
**Tip #1 – Try using an attack time around 40ms and a release time around 60ms (if it's an instrument).**

Now, these are just guidelines.

And yes, I have already mentioned these figures. But I just wanted to bring them up again.

*In no way will these settings work for every instrument.*

But it's a good starting point. You're not going to mess anything up too much with these settings.

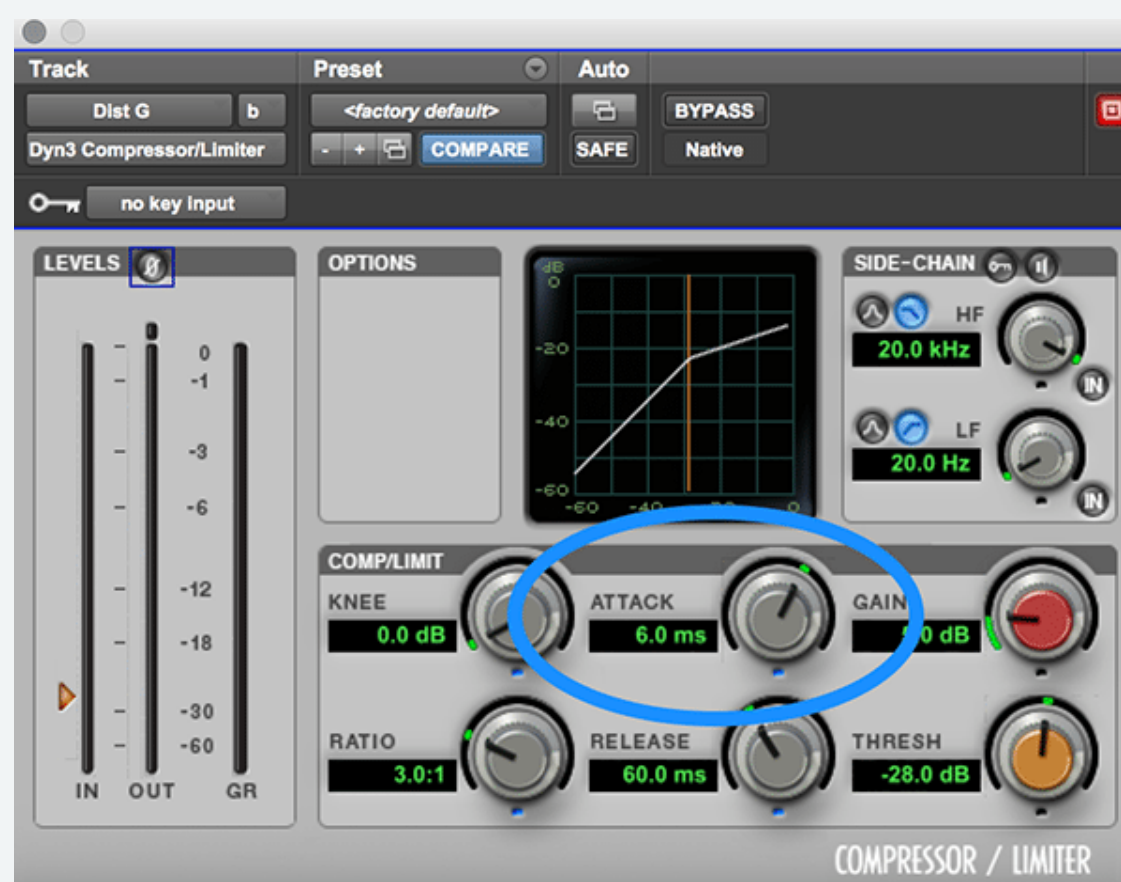


**Tip #2 - Try using an attack time of 6ms for vocals.**

You should approach vocals and voice differently when you apply compression.

Lower your attack time to around 6ms and tweak from there.

You can start with a release time around 60ms on vocals too, though!

**Tip #3 - You don't always have to use compression.**

Use compression for a reason.

Don't just use compression for the sake of it. There are two main reasons you would use compression – to control dynamics or to change the tone.

If you don't have a clear intention or a clear problem that needs fixing, don't reach for the compressor.

**Tip #4 - Stack compressors for more control.**

Let me tell you a secret...

You can use more than one compressor on a channel!

Using multiple subtle compressors in a row (especially on vocals) can sound more musical. Rather than forcing one compressor to do all the heavy lifting, each plugin does its bit.

**Tip #5 - Compress the bass guitar and kick together.**

This is a great little trick that I like to use.

The bass guitar and kick drum are usually the most heavily compressed instruments on a track. They provide the bottom end and usually anchor the song.

This trick doesn't always work. Depending on the song, you might not want to apply heavy compression to both of these instruments. You might want the bass guitar to have a high dynamic range.



BUT, if you have a song where you want the bass and the kick to be pretty consistent, try compressing them together.

Send them both to a stereo buss/aux and apply compression to both of them at the same time.

This really helps them to work together and form a solid, hard-hitting wall of bass. Awesome!

**Tip #6 – Parallel compression on drums is more subtle.**

If you want to compress the whole drum kit (not just the kick drum), you need to be very careful.

Applying compression directly to the drum overheads can quickly lead to disaster.

There is a way around this though – use parallel compression.

What is parallel compression? It's when you mix the original, uncompressed audio with the new, compressed audio.

For example, you might want to have 50% uncompressed drums and 50% compressed drums.

This sounds a lot more natural and maintains the natural dynamics of the drums. But at the same time, it makes the drums sound bigger and heavier.

Give it a go!

Some compressors have this ability built-in. If yours doesn't, it's easy to do. Send all your drums tracks to a stereo buss/aux (and maybe even the bass too).

Make sure you do this with sends, not by changing the output of the tracks. You still want the original tracks going to your master output. Then, apply heavy compression to this new stereo buss/aux. Aim for 8dB reduction or more. Slowly bring up this new 'compression' buss/aux until you can just about hear it and it sits nicely under the rhythm section.

Experiment with EQ (trying boosting the lows and highs) and different compressor settings.

**Tip #7 - Apply subtractive EQ before compression.**

Low frequencies carry a lot more power than high frequencies. They're a lot harder for a compressor to work with.

You should always put your subtractive EQ and high

pass filters before the compressor.

**Tip #8 – Pick a plugin and stick with it.**

Although compressors always have those 5 key parameters, they vary a lot in their extra features. It's best to settle on one compressor and get used to it.

The one that comes with your DAW will be more than sufficient in most cases.

Don't confuse yourself by using different plugins every time. Stick to one and learn it inside out!

**Tip #9 – Watch the meters.**

Most compressors have a graphic representation of how much the audio is being compressed.

If the compressor doesn't frequently disengage and return the audio to normal, your threshold is too low.

Watch the meter and make sure the compressor isn't constantly on.

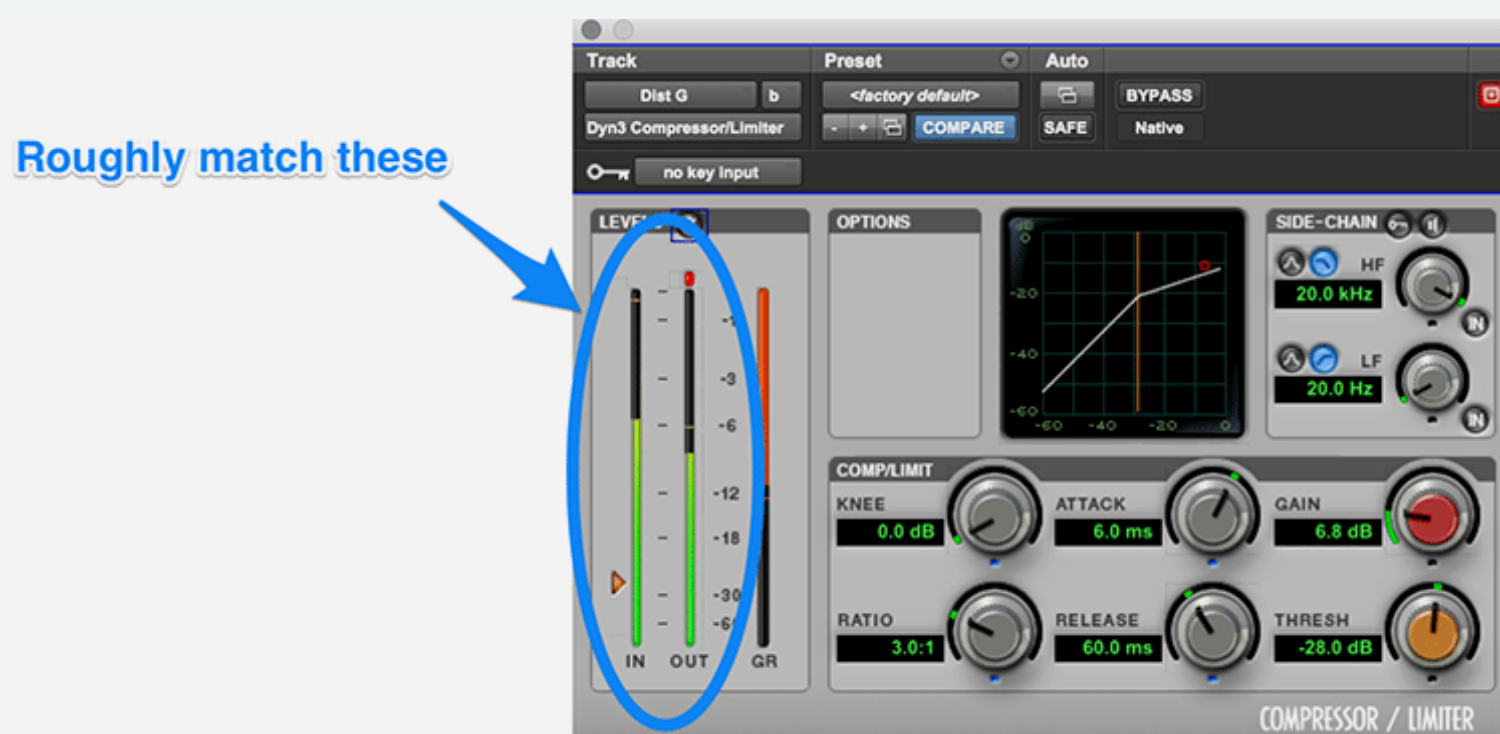
**Tip #10 – Match the output gain to the input.**

This is a common problem in mixing (and mastering)

To our ears, an increase in volume sounds pleasant. We perceive louder volume as better.

You should frequently compare your new compressed audio to the original audio to see if you have made an improvement.

If your output is louder than the original audio, you will always perceive the compressed audio as better. To avoid this, it is important to match the output volume to the input volume when you are tweaking the compressor.



Then, when you are satisfied with the changes, boost the gain if you need more volume on the track. But only after you've adjusted the parameters.

## BONUS TIP (KNEE)

Aren't you lucky! Here's a free bonus tip!

You may have noticed another setting on your compressor called 'knee'



By default, most compressors have what we would call a 'hard knee' (low number). This means that as soon as the audio hits the threshold, the compressor kicks in and reduces the gain at the ratio that we set



But what if we want the compressor to act more subtly and gradually increase gain reduction as the level of the incoming signal increases?

We can achieve this by using a 'soft knee' (high number).

Using a soft knee on vocals, piano and melodic instruments can make compression less obvious and more natural. But on a more rhythmic instrument, such as drums, you should use a hard knee.







**REVERB**

**11**

# WHAT IS REVERB?

Aren't you lucky! Here's a free bonus tip!

You may have noticed another setting on your compressor called 'knee'

Dig up any old dictionary or do an internet search and you're going to the basic definition (and understanding that most people have) of reverb:

- **Reverb Definition - an electronically produced echo effect in recorded music. Short for "reverberation."**
- **Reverberation Definition - a sound that echoes.**

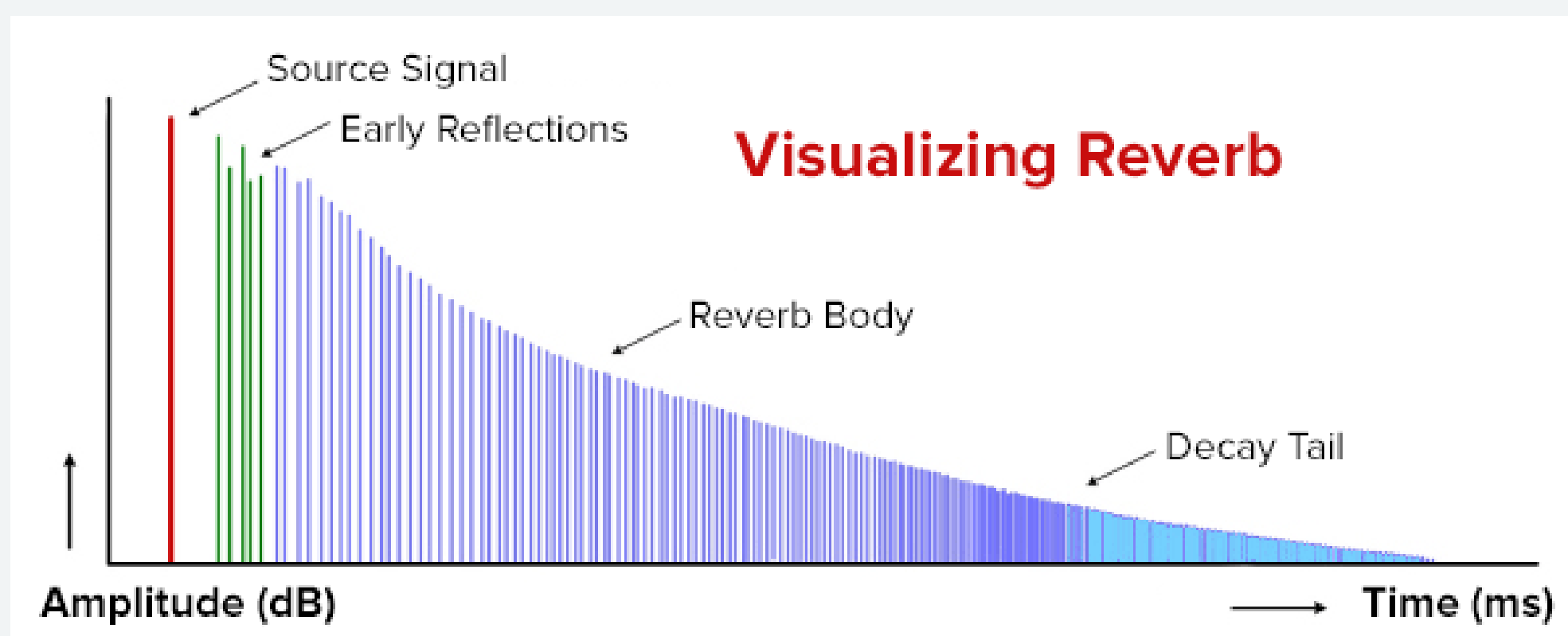
That gets us nowhere fast. To paraphrase, reverb is an electronic reproduction of an echo... The problem with this is that in the music industry, especially for a mixing engineer, there's a huge difference between an echo and reverb.

Echo has a long delay in between repetitions, like calling your name out into the Grand Canyon and waiting to here it bounce back.

Reverb is thousands upon thousands of repetitions occurring so fast that you hear it more as a smear of

sound based on the original. We're talking around less than 0.1 seconds versus several full seconds. If only reverb was that simple. Before we can use it effectively, we need to understand the three aspects of reverb:

- Early Reflections - Anywhere around 1 to 30 milliseconds
- Body - From around 30 milliseconds to forever depending on your choices
- Decay - The final tail end of the reverb sound as it drops below about 60dB in amplitude



## TYPES OF REVERB

Fortunately we don't need to design rooms (even in a simple plugin) every time we want to use a reverb. We can run to a certain preset we know has the sound we need and use it as a starting point.

This is possible because reverbs typically behave in

one of five ways, which represent the five categories of reverb:

- Room
- Hall
- Chamber
- Spring
- Plate

You can dig in and find more styles, sub-categorize them, etc. But those are your main five types.

The first three are based on rooms. These are the types of reverb you hear out in the physical world as you walk through buildings. The last two are types that mankind invented based on the vibration of either a spring or metal plate. I love me a good plate reverb!

We could go into a lengthy discussion trying to use words to explain how each one sounds. But really if you aren't aware, you can fire up your DAW and listen to each one or just wait until you're mixing again and run through them.

Being able to identify the sound of each type of reverb helps you decide which sound you want to

impart onto your mixes. Let's get to the good stuff now and talk about how to make specific decisions about reverb.

## HOW TO CHOOSE THE RIGHT REVERB

Before we even talk about the tricks of making reverb sit perfectly within your mixes, it helps if we're making the right choices from the start. Some reverbs are more appropriate than others to create certain moods. Some aspects of reverb work better with various types of arrangements.

Your main concern is choosing the right emotional quality for your mix. As an example, does the song call for something warm and natural or something metallic and dark?

Your secondary concern is to make sure you're emphasizing qualities of the reverb that don't contribute to a harsh, muddy, or spacey mix.

These two concerns are critical to achieving that professional sounding mix. If you screw this up, it's not the end of the world but you're limiting yourself. The tips and tricks we're going to introduce later can't hide bad decisions.



When choosing a reverb, there's a series of questions that are helpful in guiding your choices...

"Do I want a natural, reality based sound to emulate listening in a real room? What size and kind of room? Or do I want to use reverb more as a noticeable effect and ear candy for the listener?"

If you want to emulate a specific type of space, such as a smoky bar or concert hall, then obviously you want to choose a room, hall, or chamber (based on size first). Then you want to consider the material of the walls, which is how a lot of plugins will label their presets.

For instance, you might decide to mix a five-piece bluegrass band's song using a "Small Room - Wooden Verb" preset. That might sound like your listeners are gathered around the front porch of a log cabin as the band plays from their rocking chairs.

If you're going for more of an effect than a spring or plate reverb is going to supply you with what you want better than the more natural room-based reverbs.

# REVERB MIXING TECHNIQUES

Now that you know which reverb you're going to use, it's time to slap that sucker on the tracks and start tweaking!

## 1. LEVELS

### *HOW LOUD SHOULD THE REVERB BE?*

Generally, you're going to want to place your reverb on individual instruments versus the master output. This is how it works in reality. Each individual sound being created in a room generates its own reverb. Typically, the louder a reverb is the further back in the soundscape an instrument sounds like it is.

This is how you create a sensation of depth. Mentally visualize how you're designing the stereo field and where you want each specific instrument to sit, not only with left-to-right panning but front-to-back depth as designed by volume and reverb.

The first mistake of amateur mixers is to increase the volume of the reverb itself far too high. You must remember that all of your reverbs are going to combine into a louder overall reverb effect. Always set your reverb volumes with the full song playing, not with each instrument in solo.

The golden rule of almost every signal processing effect is to increase the volume to where you want it, and then turn it back down some. Less is more.

Dropping the volume of the reverb back by about 2 dB to 4 dB under where you like it is a nice safe spot. Subtlety is the name of the game.

Here are some general tips for each type of instrument based on their frequency response and importance to a mix:

- Your Kick Drum and Bass Guitar should be 100% dry
- Use more reverb on the toms and cymbals than the snare
- Use enough to push guitars and keyboard behind the vocals and near the snare in depth
- Lightly sprinkle the vocals with reverb, less than the other instruments to keep it up front and clear
- 

If you visualize your soundscape as if you're standing in front of a band set up like it would be on stage, this can help guide you to making decisions about how much volume your reverb should have on each instrument (and it helps with panning!).

## 2. TIME

### *BALANCING DEPTH THROUGH ROOM SIZE*

Remember the discussion about early reflections, body, and decay? This is how you can change the perception of how large the space is that your music is emanating from.

Early reflections, the first audible part of the reverb, are going to hit your ear faster if it's a small room or slower if you're designing a larger room. The body of your reverb is the main time variable.

The longer it lasts, the larger the room will feel and the more distant the instrument will seem from the listener. The decay can occur fast, making the reverb stop more suddenly, or can trail off with a long tail.

A lot of times you can't simply choose based on the size of the room you want. If your arrangement is very complex with many instruments then a large room is going to become a washed out mess.

If your arrangement is sparse, you can get away with longer reverbs. Take care to consider your arrangement as well as your desire for room size.

Because we are artificially and electronically altering each of these, we can end up creating virtual spaces that simply couldn't occur in reality. You'll know if something is absolutely whacky in your reverb or not.

Experiment with these variables to produce a pleasing and realistic sound. Getting this wrong can be worse than a completely dry mix, so take your time!

### **3. SEPARATION**

#### ***MONO OR STEREO REVERB?***

This is an easy place to create disaster for your mix. Stereo reverbs can sound so lush and full that it's tempting to use them every time on every instrument.

Once you're done setting your levels and balance, you play the full mix ready to sit in amazement at your work and realize you have a washed out, mushy mess on your hands.

So maybe you try to turn down the volumes and all you do is turn down the mush, and it's still mushy.

What went wrong?



If you're using your DAW's mixer like most of us are, this will happen for you because the reverb insert is coming in after the track's panning. Or you're sending the signal post-pan to a bus that has the reverb on it. A mono reverb will sound like it's right behind the instrument creating it.

A panned stereo reverb will have it's center of mass panned, but the echoes will fan out across the entire stereo field. And that's your mush problem.

The question needs to be "Should I use mono or stereo reverbs?" and the answer, like above with the balance, should be based on the arrangement. For instance, if I was mixing a small band I might employ LCR mixing.

I'd pan the guitar all the way left, the synthesizer all the way right, and leave the drums, bass, and vocals up the center. And then I'd slap a stereo reverb on all of them and let that wash through the empty space in the stereo field. This would sound lush and still have tons of separation.

If you have tons of instruments panned all throughout the stereo field, you can't get away with

a ton of stereo reverb and should use mono reverbs to help maintain isolation and clarity.

## **HOW TO ADD REVERB: WARNING FOR NEWBIES**

There are three boo-boo's all of us make when we first start mixing, and when it comes to reverb they can be very problematic. They are:

1. Mixing your reverb in headphones versus monitors.
2. Slapping separate reverbs on every single track.
3. Not using sends to control your reverb.

Here's the problem and solution to each of these scenarios...

### **1) CONTROL YOUR ACOUSTICS, DON'T LET THEM CONTROL YOU.**

Most of us don't start with mixing rooms chock full of a primary set of monitors, a secondary set, and acoustic treatment on the walls. We typically start with our basic computer speakers, a set of normal headphones, and a room that produces reverb on its own.

We tend to mix in headphones because our rooms are harming our ability to hear clearly more than helping.

a ton of stereo reverb and should use mono reverbs to help mRegardless, mix your reverbs with your monitors or speakers so you're setting realistic levels.

With headphones, the clarity provided by the best studio headphones can cause you to use far too little. At the same time, using monitors in a poor acoustic environment can cause you to use too much. Start with your best monitors and double check in your headphones until you find the right balance.

aintain isolation and clarity.

## **2) ANOTHER SCENARIO IS THAT AMATEUR MIXERS WANT TO MIX.**

They don't want to learn how the software works and most have never worked on a real hardware mixing board. This results in not even knowing about auxiliary sends and buses, let alone actually using them. So we end up copy and pasting our reverb plugin settings onto each track.

Reverb is a very demanding process to calculate for your computer. With too many instances of the plugin running, you risk your software freezing and losing all

of your work. Plus it's a very inefficient way of working with reverb, which leads us to the next point.

### **3) LEARN HOW TO CREATE SENDS TO SPECIFIC BUSES**

A bus is an extra track in your multitrack that takes an input from other tracks. It can take as many signals as you'd like to throw at it at any volume you choose. This is the best way to manage your main reverb.

You won't always use only one, but there will almost always be a main and it should be controlled from a master reverb bus. There's a lot of reasons why, that we'll cover in the advanced tips and tricks coming up next.

Here's the goal. We're going to take your reverb from the left-side amateur style to the right-side professional style as visualized below:



Notice that on the pro's side, the reverb never gets in the way of any of the instruments. It's in the background at a reasonable volume, EQ'd out of the way, compressed and controlled... Here's how you do it.

## **ADVANCED REVERB SETTINGS, TIPS & TRICKS THE PROFESSIONALS USE**

At this point, if you've been working along with a mix as you've read this, you're probably sitting on a better sounding reverb than you've ever produced. But it still doesn't compare to the professionals. The levels are right, the balance feels good, there's depth and separation, but the clarity isn't there.

It's still mushy, washed out, and hard to make out each sound individually. You've lost intelligibility in your mix and we're about to get it back!

### **1) PRE-DELAY - THE REVERB GODSEND**

There's an old mixing trick to increase intelligibility in an instrument or vocals by introducing it with a louder volume before dropping it down to the proper volume for the rest of the song.

It helps the listener's brain latch on to it. I don't



suggest doing that, but there's a similar trick regarding reverb that involves a feature called pre-delay.

Pre-Delay is a time based setting in milliseconds. So let's say you chose 50ms as your setting. What this does is tells your plugin to not start producing reverb for the first 50ms you'd expect it to. It waits that long before it fires.

If you set this length just past the longest attack of the guitar pluck or vocalists words, it keeps the signal completely dry long enough for your listener's brain and ears to hear clearly what's coming.



With pre-delay, the listener can then anticipate the rest without having to pick the nuances out of the wet signal. You want this, and usually 15ms to 75ms will get the job done.

Do the same on the high-end around 10 kHz and up. Suddenly your cymbals and other crystalline sounds are clear again, yet the track still has enough reverb!



Finally, you can notch out space for each instrument in the buses. As a vague example, let's say that your vocals have the most presence and intelligibility from 1.5 kHz up to 5.5 kHz. Don't hesitate to apply a wide cut of anywhere from 3dB to 6dB in that range.

The same goes for other grouped instrument buses (although I'll typically only ever use one). Remember, the reverb is welcome to shine through anywhere else but where the instruments need to. There, it needs to be much quieter.

### **3) COMPRESSION - STRAIGHT AND SIDE-CHAINED**

Yes, you read that right! Compressing your reverb can help a ton if you're having clarity issues. The opposite works too... it all depends on each different mix.

Say that you have an independent reverb set up for your vocals. Instead of EQing the amplitude of specific frequencies down, you can set up a compressor for the reverb that's side-chained to the vocals. We wrote more about sidechain compression if you need a brush up.

What this will do is quieten the reverb to some degree as your vocals are playing and then return to the original volume when the vocals aren't playing. This achieves a similar result as EQing but leaves the frequency response intact.

Alternately, you can compress your vocals (and any other instrument) but send the pre-compressed signal to the reverb.

This way the reverb will pump naturally with the amplitude of the original take, creating a sense of dynamics, breathing, and realism in the mix that otherwise wouldn't exist if you based your reverb off

of the compressed signal.

You can even combine this breathing version with a compressed version as a form of parallel reverb.

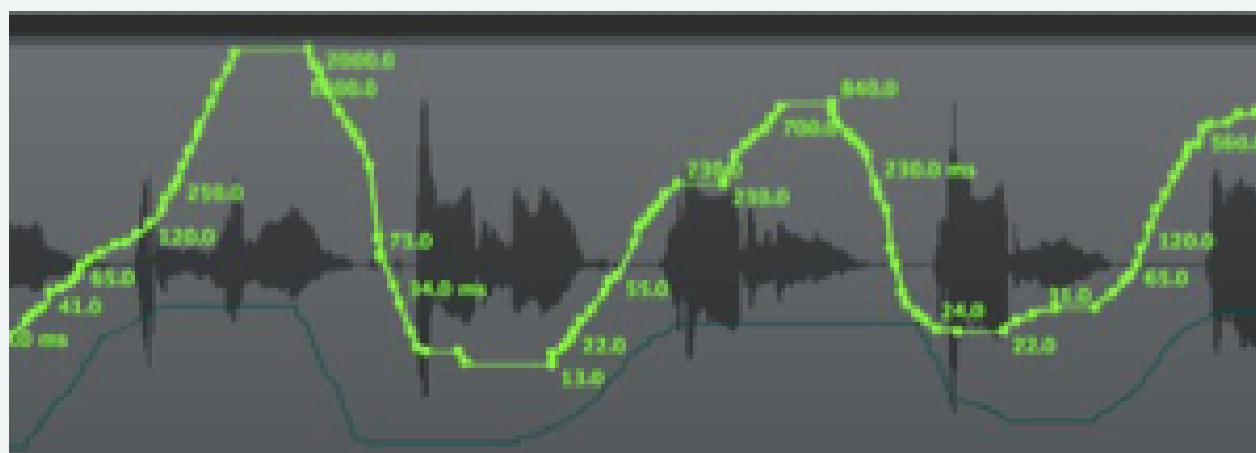
Lastly, you can compress your reverbs straight so that they ride a steady amplitude that doesn't distract or get in the way. If you do this, you'll want to use EQ on it as mentioned above.

In general I don't find myself needing to straight compress reverb because I apply reverb to the compressed source signals anyways and never have it loud enough that it's a problem.

Your mileage may vary. Definitely experiment and develop your own style. Watch out though, depending on your compression settings you'll start to affect the feel of the decay tail.

## **4) AUTOMATION - MANUAL PRECISION OF REVERB VOLUMES**

The last pro-tip is to apply an automation track to your reverb's volume. This may be the wet/dry knob, which does the same thing.



What you can do is manually draw in your automation parameters with your computer mouse to selectively drop or raise the volume of the reverb at specific parts of the mix that are giving you trouble.

This also provides a way to use it as ear candy to accentuate certain vocal phrases, for instance.

## **5) OTHER FUN EXPERIMENTS - REVERSE REVERB, GATING, ETC.**

Now that you have a full handle on using reverb in a professional manner, you can start to break the rules and play around. Try bouncing the reverb send, reversing it, and allowing the decay tail to become the early reflections.

This can be a neat effect on drums (especially snare) and can be used for more spooky genres and songs. You can gate your reverbs as well so that they suddenly cut off, making a drastic impact on your listener as it is unnatural and will catch them off guard.



Reverbs, delays, echoes, and other time-based effects are the key elements to what I call "ear candy", creating special moments in songs that really stand out. Go as far as to place flangers on your reverb! Experiment!

Reverb is a must in 95% of mixes, I would guess. You can't escape it, and your choices are to do it wrong or do it right. You might produce the world's most perfect mix ever and move on to reverb and destroy it. You might know how to mix vocals like a god and then butcher it with reverb.

Knowing which reverbs to use, how to design the virtual rooms, and then how to apply a pre-delay and EQ are the three aspects of reverb that separates the amateurs from the professionals.

You've either got a muddy mess or a clear and beautiful mix. You can't escape mixing with reverb, so make sure you know how to use reverb properly or you're doomed before you begin.



**DELAY**

**12**

# WHAT IS DELAY

Delay is an audio processing technique that records an input source to a storage medium (like a reel of tape or a hard drive), then plays it back after a user-defined period of time.

Delay can be a great way to add space to a track without taking up much room in the mix. Much like reverb, delay can be used to push a track back in the mix and create the illusion of depth. It can also be used to make tracks sound wider by using a stereo delay on a mono signal, or panning the delay to the opposite side of the mix

## DELAY PARAMETERS

- Type/Mode – Changes the delay type.
- Delay/Time – The amount of time between each repetition/tap.
- Sync – Enable to sync the delay time to the BPM setting in your DAW.
- Feedback – How long the delay lasts for after the first tap.
- Mix – For adjusting the level of the delayed signal in relation to the original.

- Low/High Cut – Filters for cutting low or high frequencies from the delayed signal.
- Saturation – Applies analogue style harmonic saturation to digital delays.
- Modulation – Changes the tone of the delayed signal over time.
- Width – Widens the delay across the stereo field.

## THE DIFFERENT TYPES OF DELAY

As with reverb, there are many types of delay. These are the most commonly used delays:

- Slapback: quick, punchy delay that mimics a large room.
- Doubling: similar feedback time to slapback delay but with a shorter delay time. Basically, it sounds like it doubles your signal.
- Loopback: works like a typical delay except it doesn't have decay. It will loop your signal continuously unless you manually adjust or remove it.
- Chorus: copies the original signal like a typical delay. Then it alters the frequency before adding it back in.
- Flangers: basically, a chorus effect but with a shorter delay time. Phaser: sounds similar to a

chorus or flanger effect. But it uses all-pass filters instead of actual delay.

There are many other types of delay, both digital and analog. But this list includes the basic ones you need to know about.

## 5 TIPS FOR USING DELAYS

### 1. PLAY WITH VARIOUS NOTE DIVISION OPTIONS TO ADD RHYTHMIC VARIETY.

With almost all delay plugins, you have the option to work with either note divisions or milliseconds. If you sync the delay effect to the tempo of your session, parameters that are measured in milliseconds will transform into note divisions instead.

When it comes to rhythmic variations, you can add to any audio or MIDI recording with a delay plugin, the possibilities are endless. Those possibilities multiply when you work with a stereo delay

If you're working with a stereo delay to create a ping pong effect, you can set the left channel to play back echoes in quarter notes while keeping the right channel in eighth notes.



Almost all delay plugins come with additional note division options like dotted notes and triplets. Some might even give you the liberty to control the groove of the delay, with a range from shuffle to swing.

Whether it be vocals or tribal drums, delay effects can be applied to a variety of sounds and instruments to add rhythmic variation and excitement to any arrangement.

If things get a bit out of control in the process, remember that you always have the feedback parameter to turn to. Keeping the feedback knob down will ensure that echoes won't feed back into each other and multiply over time.

Many DAWs come with flexible delay options these days. But if you're looking for a third-party alternative, you can't go wrong with Soundtoys' EchoBoy.

## **1. USE A TAPE DELAY IN PLACE OF A CHORUS OR DOUBLER FOR A MORE NATURAL AND WARM THICKENING EFFECT.**

When it comes to adding more body to any sound, many producers reach for a chorus plugin.

Sometimes in vocal production, a similar doubler effect is preferred. Yet while both of these options are time-based effects that function in a similar way to a delay effect, they tend to alter the sound a lot more in comparison.

With a tape delay plugin, on the other hand, you can thicken your sounds without creating the effect of “hearing double” on the listener. This method can work on just about anything you want to fatten up a bit, from vocals, to snare drums and even synth pads.

The trick is to make sure that you don’t sync the tape delay plugin of your choosing to the tempo of your session. Try measuring the echoes in milliseconds instead. The second step you need to take care of is turning the feedback knob all the way down, so that the echo occurs only once. From there, you can go ahead and adjust the delay time and mix according to your taste! But remember that

the longer the delay time is, the farther away the echo will be from the original sound.

In this category, Waves J37 Tape Saturation is my personal favorite.

### **3. ADD DEPTH TO ANY SOUND WITH FEEDBACK.**

Turning up the feedback control in a delay plugin can be a scary thought, since it can lead to constantly multiplying echoes that know no bounds. But that exact fact is what makes it so fun to play with for sound design purposes.

Next time you're working with a monophonic instrument or a vocal part with a simple melody, try automating the feedback parameter in your preferred delay plugin to deepen the sound.

It is worth noting that some delay effects come with a freeze button, which allows you to take this concept one step further. As soon as you hit the freeze button in a delay plugin, it will capture the sound in accordance with the delay time of your choosing and play it back indefinitely. This is a favorite strategy especially among ambient electronic composers who love building massive soundscapes.

There are a lot of stock delay effects that can serve you well in this area, but if you're looking to mix things up, Soundtoys' PrimalTap could be a good starting point. Though Valhalla's free offering, Super Massive, doesn't come with a freeze parameter, the delays it generates are indeed massive and stunningly gorgeous.

#### **4. USE DELAY AS A MOMENTARY EFFECT TO MAKE YOUR PRODUCTIONS MORE INTERESTING.**

A momentary effect can be any audio effect that occurs once or twice throughout the track. Delay in particular is used so often in this manner that it has a special nickname: delay throw.

Delay throws are especially popular in vocal productions, but don't let that stop you from experimenting. Sometimes a delay throw with a funky groove you put on a snare or tom can work as a fill or add some variation that's lacking in a beat.

I find that hybrid effects with analog character tend to work best for delay throws. Waves Audio's H-Delay is one that I usually turn to first.

## 5. CREATE UNIQUE TEXTURES WITH GRANULAR AND PITCH SHIFTING DELAYS.

Granular or grain delay takes its name from granular synthesis. It grabs a tiny section of the performance and plays it back.

Most granular delay effects come with pitch shift controls, which opens the door not only to some sonic but also compositional surprises. Imagine adding contrapuntal hints on a single melody or even creating background vocal harmony textures underneath a lead vocal performance.

Those are some of the more obvious strategies but a granular delay can serve well in situations where you're dealing with hi-hat or shaker patterns that feel too stiff. Pitch-altered echoes can liven up just about any light percussion loop. Take it far enough and granular delays might soon become your go-to for creating glitch beats, too.

My personal favorite in this category is Soundtoys' Crystallizer. However, free options from SoundHack, Pitch Delay and Bubbler, should not be overlooked. For electronic music producers, Glitchmachines' free delay plugin Hysteresis could be another inspiring choice to consider.





**SATURATION**

**13**

# WHAT IS SATURATION?

Audio saturation is the essence of what makes analog hardware sound musical and pleasing.

Driving sounds through tape, tubes, transistors, and circuits have long been an essential ingredient in great-sounding mixes.

Saturation is a subtle form of distortion that adds pleasant-sounding harmonics. The effect originates from the analog days when audio recordings ran through various pieces of hardware. Mix engineers discovered they could overload magnetic tape machines, tube amps, and transistor-based preamps to create a type of “soft-clipping.” This technique gave recordings sought-after qualities that sounded pleasing.

To this day, artists and engineers use saturation to add presence, character, warmth, edge, cohesion, and more. From subtle to extreme, saturation is an integral part of amazing mixes.

# TYPES OF SATURATION

While some studios still have analog equipment, the industry has shifted to the digital realm. There are

several types of saturation plugins that emulate the imperfections of analog hardware. However, the most common types are tape, tube, and transistor.

## 1. TAPE SATURATION

Tape saturation plugins emulate the sound of audio recorded through tape machines. They introduce odd order harmonics, subtle compression, and non-linear shifts in frequency response.

Tape saturation also rolls off high-end frequencies and creates a small boost in the lows. Moreover, it rounds off transient peaks, creating a form of compression that smooths out the signal.

Many describe tape saturation as sounding warm and punchy. This type of harmonic distortion works great for adding dimension, fatness, and depth to your mix

## 2. TUBE SATURATION

Tube saturation plugins emulate the sound of audio driven through tube amps. They introduce even order harmonics. This type of harmonic distortion also adds a subtle form of compression. However, when pushed hard, tube saturation can have an

aggressive edge.

Many describe tube saturation as sounding warm, musical, and punchy. This type of harmonic distortion can also increase perceived loudness, dimension, and fatness

### **3. TRANSISTOR SATURATION**

Transistor saturation plugins emulate the sound of audio driven through transistor-based circuits. You can also achieve this sound by overloading input levels on various hardware and plugin emulations. Transistor saturation introduces odd order harmonics and a form of “hard clipping” compression. Depending on the device, this type of harmonic distortion can have a subtle or aggressive sound.

Many describe transistor saturation as sounding fuzzy, gritty, and textured. Moreover, its impact on transients and higher harmonics can make sounds less punchy and musical when pushed hard. Whereas, subtle settings will give you a smoother tone.

# HOW AUDIO SATURATION CAN IMPROVE YOUR MIX

Audio saturation plays a significant role in music production. Applying saturation to single tracks, bus groups, and the master channel will improve your mix. Even more, it adds that desired analog vibe!

Various flavors of saturation generate harmonics and apply subtle forms of “soft-clipping” compression. This unique effect makes sounds fuller, punchy, and louder. Saturation also adds depth, presence, character, color, and warmth. Moreover, it does an excellent job of “gluing” groups of sounds together. This technique helps create a cohesive mix when applied to bus groups and the master.

## APPLYING SATURATION

### 1. DRUMS BUS

Applying tape saturation to your drum bus helps “glue” all your drums tracks together. It also helps tame rogue transients and softens high-end harshness. This method is a great way to inject character and excitement. Moreover, it will add subtle “punch” to help your drums cut through the mix.



## 2. BASSLINES

Injecting tube saturation to your bass parts will add power and fatness. Driving your bass through harmonic distortion will also add mid-range to help it cut through the mix. Experiment with different types of saturation. They will enhance your bass big time!

## 3. VOCALS

Subtle amounts of harmonic distortion will make your vocals pop! Apply saturation to tame wild transients and make thin or dull vocals fuller. It also does an excellent job at warming harsh sounding vocals. It's the secret ingredient to great sounding vocals.

## 4. SYNTHS

Virtual synths often sound too clean and digital. Saturation will bring static synths to life with some analog warmth, fatness, and character. Try some tube saturation to add harmonics and grit. This method will increase perceived loudness and bring your synths upfront.

## 5. MASTER

Applying tape saturation to your master adds a cohesive, analog sound to your mix. However, be gentle

with the amount. A little goes a long way.

Inserting a console emulation plugin to the master is another common technique. This move simulates running your mix through an analog console. It also works great on single tracks. Also, try adding channel strip or preamp plugins to several tracks. This method creates a cohesive, analog vibe across your entire mix.

## CONCLUSION

Infusing your mix with authentic analog character is an excellent way to enhance your music. Also, trying different saturation types across your mix will give you better results. For instance, try using tape on drums, tube on vocals, and transistors on synths. And, remember not to go overboard, or you can cause more harm than good.



# MONITORING YOUR MIX

# 14

One of the hardest parts of mixing is knowing when to stop. Most engineers have an innate desire to tweak things. It's part of what makes us good at our jobs—we love to take things apart and tinker with the pieces.

But too much mixing can actually be a bad thing. Eventually, you stop improving the mix and start making it worse. That's why it's so important to stay focused while mixing to ensure you don't lose sight of the big picture. Check out our seven-step checklist for making sure your mix is finished.

## **YOU CAN HEAR EACH INSTRUMENT CLEARLY**

On a basic level, the goal of mixing is to make sure you can hear each element clearly. Before pressing print on your final mix, listen closely to make sure you can hear what each instrument is playing at all times.

When two instruments occupy the same space in the frequency spectrum, it can make it difficult to hear them clearly. For instance, if you're having trouble hearing the kick and bass, carve out space in different ranges for each instrument. Try cutting the

lows on the kick and the low-mids on the bass to make room for both instruments. For even more separation, boost the low-mids in the kick and the lows in the bass.

## **YOU CAN UNDERSTAND EVERY WORD OF THE LYRICS**

Arguably, the vocal is the most important track in any song. It should be the focal point of the mix, which is why it's important to make sure you can understand every word.

Start by checking the level—the vocal should be the loudest track in your mix. If you notice that the vocal occasionally sounds too quiet or too loud, try using a compressor to maintain consistent levels. Use a slow attack and a fast release for a natural sound, and apply 3-6 dB of compression with a modest ratio.

If you also happen to be the person singing on the track, after you get the vocal level sounding right, bring the fader up by 1 dB. Due to the way that we hear our own voices, vocalists tend to bury the vocal when mixing. Remember, someone who has never heard the lyrics before should be able to understand every word!



## THE MIX IS GLUED TOGETHER

While it's important to have separation between each element, your mix should also feel cohesive. To make sure your tracks don't feel disjointed, use a bus compressor to “glue” all of the instruments together. Try using a slow attack and fast release settings with very mild ratios like 2:1, and as little as 1-2 dB of gain reduction.

If you find that this throws off the balance of your mix, try using a multi-band compressor to tame any frequencies that stand out. You may find it useful to compress some frequency ranges more aggressively than others. This approach can be great for making multiple tracks sound like they were recorded in the same space together.

## THERE ARE NO TECHNICAL ISSUES WITH THE MIX

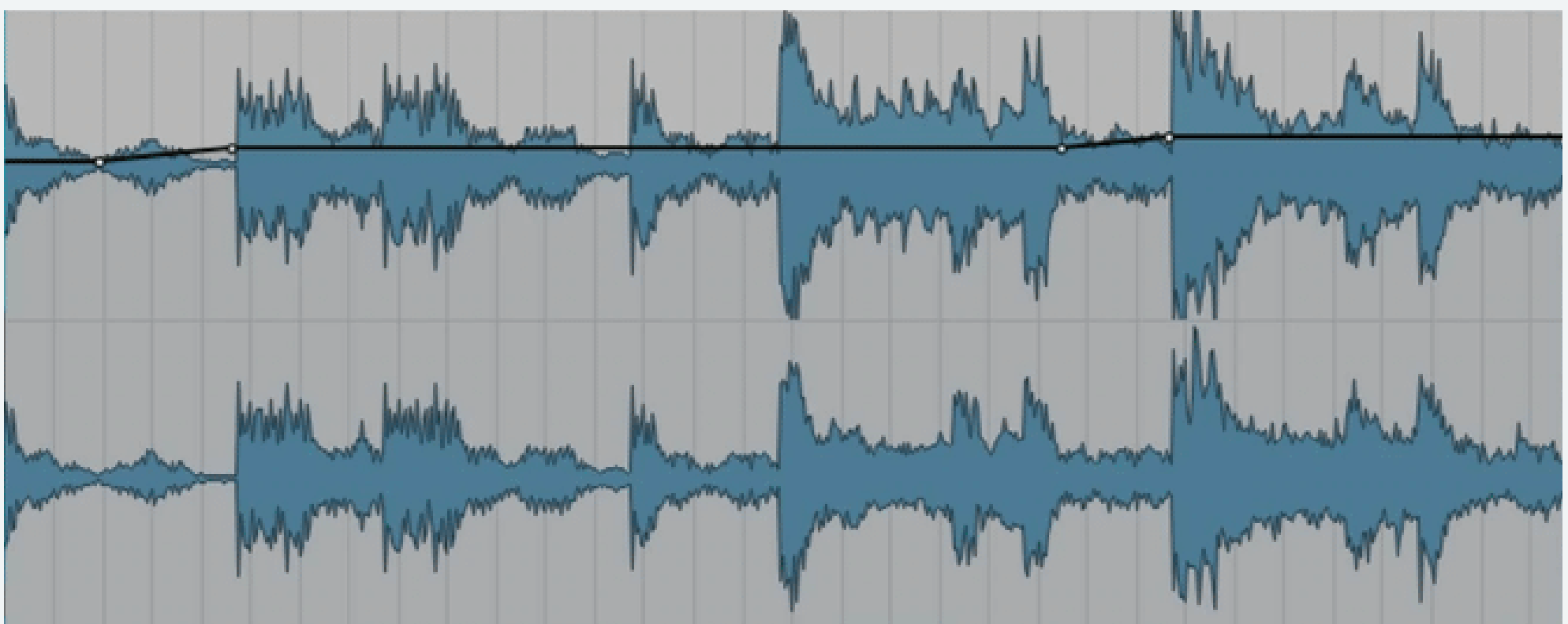
When the vibe is right and everyone in the studio is excited that you're nailing the mix, it can be easy for subtle mistakes to slip through the cracks. And while it may not seem like a big deal, those tiny issues become amplified during mastering.

# IT GROWS AND CHANGES OVER TIME

Songs that grow over time tend to be more dynamic and interesting. They often start out quiet with the intro, add in a few new elements for the verse, and pick up steam in the chorus before crescendoing at the bridge.

Too much compression can cause your mix to sound flat or stagnant. If you're going for a punchy sound but still want your mix to sound dynamic, you can automate the level of the mix bus to add energy and excitement throughout the song.

Identify the loudest and quietest moments in the song and emphasize the changes in level during these sections. Bring the verses down 1 dB and the choruses up to really grab the listener's attention.



## IT SOUNDS GOOD ON MULTIPLE SOUND SYSTEMS

Once you get the mix sounding good on your studio speakers, you may think your mix is finished—but it's only just begun. The true test is to make sure your mix sounds good on every system. When mixing in untreated rooms (like most home studios), frequency build-ups can trick you into thinking your mix sounds better than it actually does.

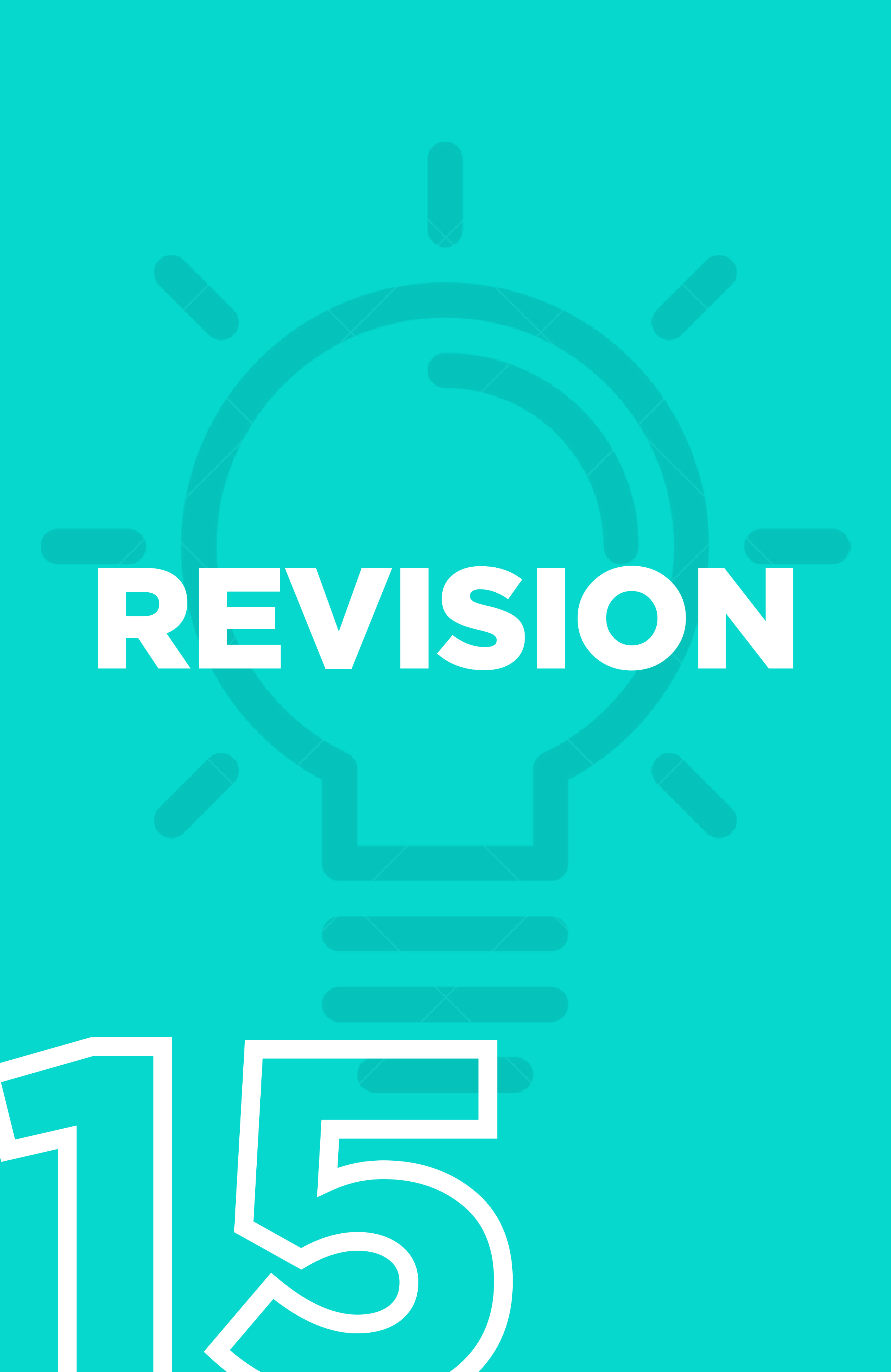
For instance, if your room has a frequency build-up around 80 Hz, you may cut that frequency out of the kick and bass. But as soon as you play it on another system with a subwoofer, it sounds weak and thin. When you can't trust what you're hearing, it becomes very difficult to make critical mix decisions .

Grab a pen and paper and take notes as you listen to your track on a few different systems, including headphones, small speakers like your cell phone or laptop, full-range speakers like a hi-fi system, and your car stereo. Use your notes to guide you through the final steps and refine your mix until you're happy with the way it sounds on each system.

# YOU LOVE THE WAY IT SOUNDS

At the end of the day, the most important thing is that you're happy with the way the mix sounds. A great mix should make you have an emotional reaction to the song. It should make you want to sing, dance, cry, or jump into the mosh pit. That's my marker for knowing when a mix is finished—as soon as I find bobbing my head and myself singing along, I know I'm getting close.

Use this checklist the next time you're mixing to help make sure you've covered all of your basics and don't over-analyze your mix.



**REVISION**

**15**



Okay, now let's do a quick recap of what you learned of far.

In the first chapter you learned what mixing is, how our mixing environment and mixing gears like studio monitors, headphones can affect our mix.

Later than we learned about the problem we have to face if we are working in an untreated studio and how mixing with studio monitors in an untreated room is not an ideal options.

In the third chapter we got a detailed explanation about how to set up your Studio Desk, where to place your Monitors, and at what height the monitors should be placed.

Also, we looked up in detail about where to place your desk, and most important where NOT to place your desk in the studio in order to avoid resonance and all the other things you could possibly do if you can't afford or don't want to do acoustic treatment of your studio.

In the forth chapter we guided you on how to do acoustic treatment of your home studio or studio, what are the things that are necessary to do before applying acoustics.

What are the most problematic areas of your studio and what can you do to improve them.

Also the types of sound-absorbing or diffusing equipment you could use in some areas and the things to consider before buying any acoustic panels, bass traps, and diffusors.

From this fifth chapter, we dived into the nitty-gritty of mixing, here we learned how to prepare for mix, what things to consider before you hit the mixing session in detail.

We did all the stuff that look less important, but when they add up together, they create a huge difference.

Also here we learned how to check and edit your samples so that we can avoid any annoying pops or clicks which often are a result of badly recorded samples.

We'll also saw what is an Effect bus or Return Track, why to use them more often than using effects as an insert. And most importantly, how to create an Effect Bus and make them ready to use.

In the looked upon how can we get started, what are the ways to get you in the ballpark of mixing, how to use Balancing tricks to get you at a good starting point.

And Further We Learned about each aspect of mixing i.e. Panning, Equalisation, Compression, Reverb, Delay & Saturation.

And at the final chapter we learned how can we monitor our mix so that we can be sure about a mix is finished

I hope you'll use the tips I've shared with you throughout this book to make an exciting mix. I'd love to hear about your mixes and productions

Submit Us Here: [info@midisic.com](mailto:info@midisic.com)

# Thank You!

Before you leave and try out all these tips I just wanted to say “thank you!” It means a lot to me that you took the time to read this far.

There are so many books on mixing and audio so I’m honored you picked mine.

If you enjoyed this book and loved what you read, please leave a review on your Social Media stories and Tag @midisic. That’s honestly one of the best ways to let others know how valuable this information is. I really appreciate it