Mixing Concepts PDF

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Foreword

This is a collection of all the techniques demonstrated in my Mixing Concepts Series.
There is no particular order to these, you can jump around as you like.

The descriptions are kept as minimal and concise as possible, what you make of these techniques will be up to you, only after trying them in practice will you really know them.
Mixing is not a science, it is a craft and an art.
There is no theory that can tell you when one or the other alternative will always sound better. Get to know your tools, get to know their sound, develop your taste and intuition.

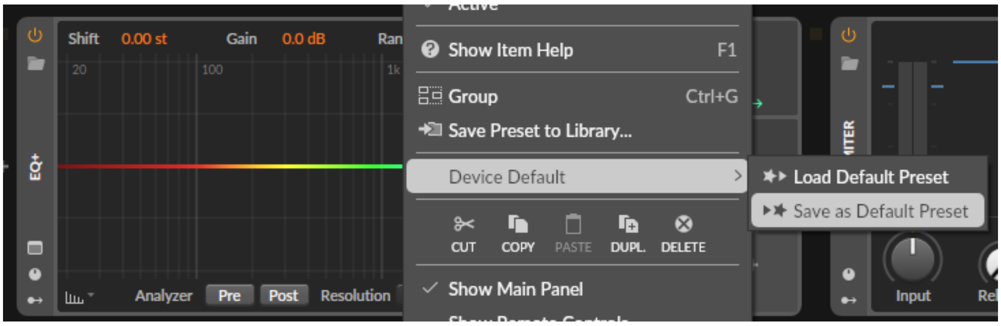
Check out the youtube series for video and audio demonstrations.
Thanks a lot for your support!

1. Turning Processing on/off

After you set up each effect, turn it on/off and carefully listen to the difference it makes. Sudden differences are a lot easier for our ears to evaluate than gradual ones you experience while e.g. tweaking a compressor or EQ. Is the effect actually improving the sound? Or is it making it worse? If it’s the latter, immediately remove it.
A nice side effect of this workflow is that you gain a lot of ear training and build your ability to imagine what certain effects might sound like on a given signal.

2. EQ without distracting Visuals

The frequency analyzer on your EQ can often be misleading. When provided with both visual and auditory information, we often trust the visuals over our ears. In everyday life this is useful, in mixing it is not. What looks odd on an analyzer and what sounds odd will usually not be the same thing.
By disabling the analyzer in your default EQ preset you will force yourself to listen first. If this makes you uncomfortable and less confident in your EQ choices, this is all the more reason to give it a go. You will get used to it after a few hours.
If you absolutely have to, you can still turn the spectrum analyzer on again later. But make sure that whenever you load in an EQ, it is first turned off, so the first thing you go off of are your ears, not your eyes.



3. Custom Gates with MCompressor

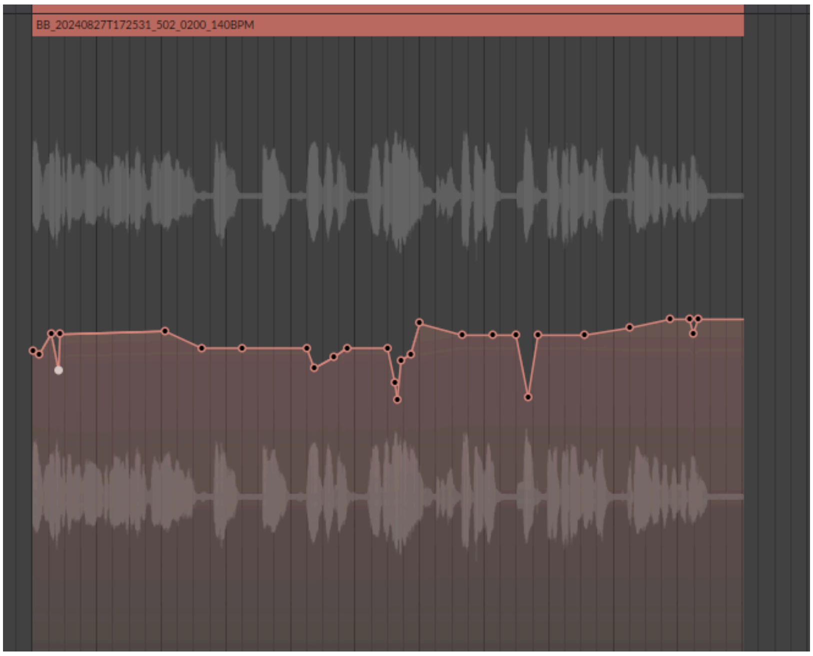
Using the free plugin MCompressor, we can really fine tune the transfer function of our dynamics processing. My most common use case for this is creating more specific gates.
Your default gate will probably use a transfer curve like the left image.
But maybe what your sound really needs is something more nuanced, like the example on the right?

See my youtube video on transfer functions if you're unfamiliar with that idea and how to make use of it.

4. Automate instead of Compressing

A dynamics processor, e.g. a compressor, is really just a tool that automatically adjusts a sound's volume depending on some kind of transfer function.
We can however also just adjust the volume of a sound manually and often this will yield cleaner, more

transparent results. The reason we don't always just draw the volume curve by hand is often merely because it would be a lot of work.
But on sounds where it really matters, it may make sense to not rely solely on a compressor, but to take matters into our own hands.
Here are two examples, one of manually "compressing" a vocal and one of manually "transient shaping" a snare drum:



Make sure to check how your DAW responds to very quick automation and whether it makes a difference if you use clip gain or automate fx parameters.
Automation of VST plugins and insert fx often only updates every N samples, which can matter for very quick gain changes, like the second example of the short transient boost.

5. Trial and Error

It is often not immediately obvious which chain of effects is the best way to process a sound. If you have multiple ideas, just try them all and compare.

In Bitwig, we have a tool for this, the FX Selector, in other DAWs you can simply duplicate a given track and create alternative versions of it. This is also how we can choose which clipper, limiter, resonance suppressor etc sounds best on a given signal.

For very subtle differences it can even be sensible to do this comparison blind, e.g. shuffling up the options and picking your favorite not knowing which is which.

6. Just swap the drums

Sometimes the best way to mix a kick drum is just to pick a different one.
When you find yourself having to do a lot of processing to your kick sound, it's worth a try to see if simply swapping it out for a more appropriate sample won't get you better results.
I tend to keep my kicks in midi until the mixing stage and just try out a couple of alternative options. This can prevent a lot of headaches.

7. Synthesize bespoke Kicks

If you're somewhat comfortable with synthesizing kicks, it can be fastest to not even try to find the right sample, but to simply synthesize a bespoke one. As a lesson from KOAN Sound made me realize, a kick drum can only be judged in the context of a mix. So during mixing, it can make sense to go back into your kick synthesizer and tweak it to better sit within a given context.
Adjusting pitch and volume envelope of a kick drum will often go further than any amount of fx processing later down the chain.

8. Arrange to make your life easier

Mixing is an extension of arrangement. During arrangement, you first make the choice which sounds you place where in the frequency spectrum. If you're careful during this phase not to create conflicts, you won't have to struggle against them later during mixing.
Sometimes simply moving an instruments part up or down an octave can spare you having to do a lot of tricky EQ work.
As you gain experience mixing, you will find yourself often regretting arrangement choices you made earlier. This regret will be a great teacher to prevent yourself from creating the same problem in the next song you write.

9. Frequency Specific Sidechain

Two sounds that exist in the same frequency area at the same time can mask eachother, make eachother difficult to hear.
Using spectral dynamics plugins like oeksound soothe, tdr speccraft, trackspacer, fuser, pure:unmask and countless others, we can make one sound duck out of the way in certain frequency areas, whenever another sound occupies that range.
This can sometimes be a better choice than simply making room for the prioritised sound A with static eq, because when sound A is not playing, sound B will not be ducked in those ranges and can still occupy them.

10. Two ways to EQ

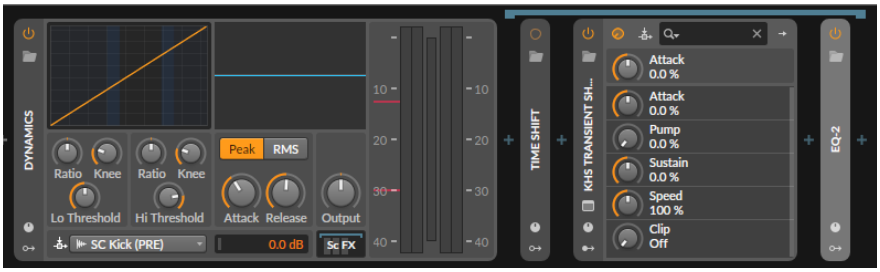
There are two philosophies I think about when it comes to EQ and in different situations either may get you better results than the other:

1. EQ as a tool to shape a sound, to change its frequency balance into the shape you want it to take. For example; this sound lacks high end → I will boost the high-end.

2. EQ as a tool to mix a sound with itself, to bring out the best of a sound. In which frequency areas does this sound sound nicest timbrally? I will boost those. Which frequency areas are less interesting/pleasant sounding? I will cut those.
For example, I may want more high end, however the high end frequency content of this sound may just be noise. Boosting this noise won't really sound that great, i should really turn it down and boost the areas where the sound shines instead. I can then add high end using other methods, e.g. by reshaping a different sound or adding in a high frequency layer.

11. Pre-Processing Sidechain Sources

When using drum sounds to trigger a sidechain compressor, we can often get more control over how the compressor reacts to them by not using the drum sounds themselves as a sidechain source and instead using a dedicated copy we create.
We can then use processing on this copy to guide the sidechain compressor's reactions, e.g. rebalancing the sound with EQ to adjust how sensitively the compressor reacts to certain frequency areas, or transient shaping the sound, to make the compressor react more quickly than it might by simply turning the attack time down.
In Bitwig we would archieve this by processing our sidechain source in a compressors sidechain fx:



In another DAW we might simply choose to create a dedicated copy of our Kick Drum Track, whose sole purpose is setting off a sidechain compressor. We can do any processing we want to shape our compressor's reaction on this copy, while keeping it muted so it doesn't affect the mix in any other way.

12. To Clip or To Limit

Clipping and Limiting are used for a similar goal, to reduce the peak value of a sound. To choose when either is appropriate, these are the things I keep in mind:

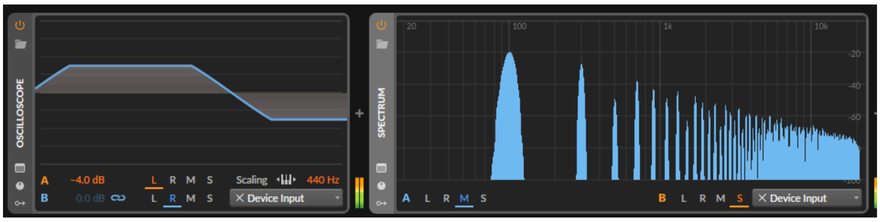
A clipper will distort more, it will add harmonics. On very short overshoots or noisy sounds this can sound very transparent, but this distortion becomes more apparent, the more sustained and tonal the clipped part of the sound is.

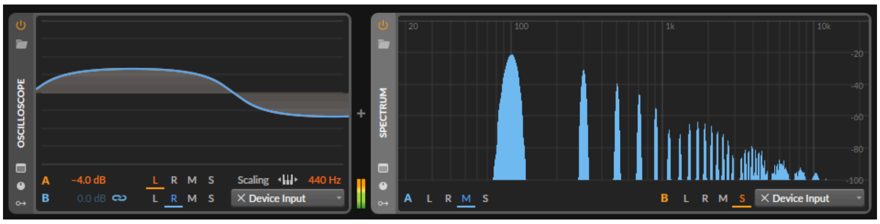
A limiter will distort less, because it doesn't just turn down the sound immediately, but turns down the sound with a release time. This prevents the waveshaping distortion of a clipper, but leads to more of the sound being turned down, so an overall quieter signal with potential pumping artifacts.

13. Soft vs. Hard Clipping

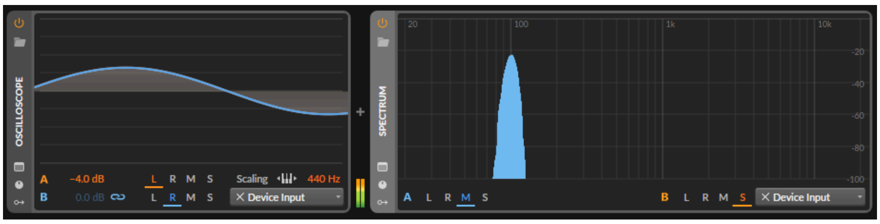
Hard Clipping can create sharp corners in a waveform, which results higher harmonics being added.

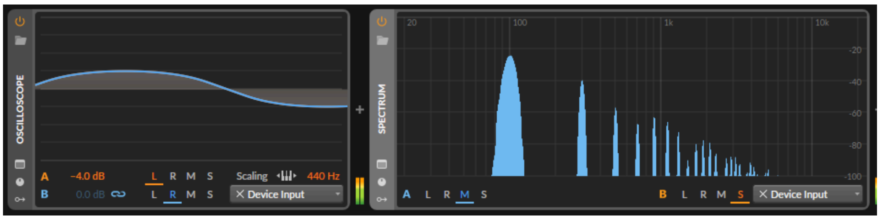
Soft Clipping will round those corners off, which means the harmonics generated are weighted more towards the lower end of the spectrum and can sound less harsh. However, because the soft clipper smoothly transitions from not clipping at all to hard clipping with its knee, it will affect more of the signal, including quieter parts that a hard clipper would leave alone entirely.
Here you can see a loud test signal being hard clipped vs being soft clipped:





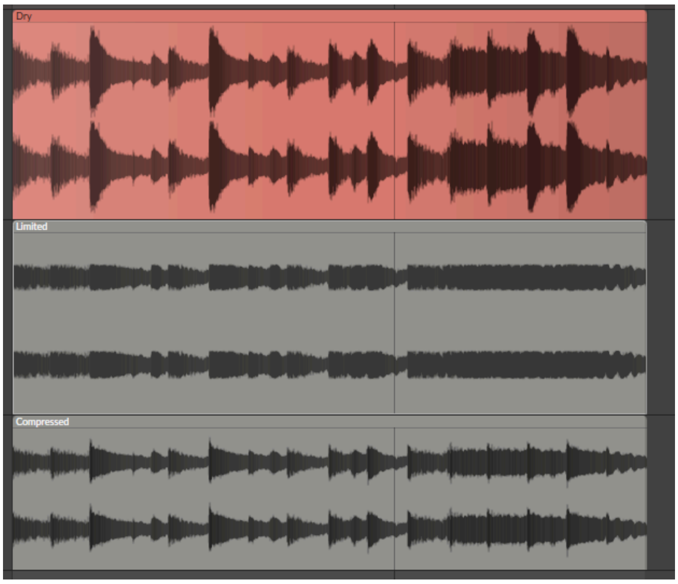
Here however is a quieter test signal being hard clipped vs being soft clipped. Notice how the hard clipper leaves the signal untouched, while the soft clipper still affects it:





14. To Limit or to Compress

Limiting and Compressing are two different ways to control the volume of a sound.
The big difference between a limiter and a compressor with infinite ratio is the attack time. In fact limiting is just a special case of compression.



Here you can see two examples of a processed drum break, one limited and one compressed, at the same threshold.
The compressor had some attack time, so it let through some peaks where it was not able to reduce the volume fast enough to ensure the sound won't exceed the threshold, while the limiter adhered to it very strictly.

If what you care about is the amplitude of the waveform, the objective measure of the sound never exceeding your threshold, limiting is clearly the way to go here.
If however your priority is more that it never exceeds a certain *perceived* loudness, the compressor has multiple advantages.
For one, it still preserves the transients, by letting them through a little bit. While the limiter completely eats up the snappiness, the compressor keeps it intact.
Another advantage is that the compressor not reacting as quickly will result in less distortion.

So when controlling a sound's volume, you should ask yourself what you care about more:
Objective peak values & the sound never exceeding them,
or perceived loudness & the volume being controlled, but not at the expense of transients and distortion.
There is no better or worse here, the right choice depends entirely on the context and your goals.

15. EQ can't add what isn't there already

As hinted at in chapter 10 already, EQ cannot create new information. It can only ever turn up or down what is already present in a sound. No amount of boosting the high end on a 40Hz sine wave will create high end information out of thin air. At best you'd bring out unwanted digital artefacts.

Analogously, when you want to add information in a certain frequency area, you must first ask yourself what a sound has to offer in this area. If there is no useful frequency content present in the sound already here, you will need to reach for other tools (like saturation, frequency shifting,...) to get the information you want to where you want it.

16. Remove everything you don't need

A very simple guiding principle I use when mixing is that I remove everything that isn't needed, every bit of information that does not contribute to the musical idea.
The assumption here is that space in a mix is limited and everything that is taking up space needs to somehow justify it.
Using tools like gates, volume automation, filtering and others I try to trim back every sound to its essentials. What exactly this means will depend on your personal tastes. Whether you're aiming for a squeaky clean, focussed, clinical sound or if you consider background noise and space to be vital to your musical vision.

17. References to calibrate, not imitate

There is a somewhat mislead idea floating around in producer spaces that the goal of reference tracks is to match your mix to them, to use them as a blueprint to exactly copy. Under this assumption I can understand how some might be reluctant to use a reference tracks, after all we want to realize our own unique vision, not just copy someone else's.
However, I think a much better way to think of reference tracks is to see them as a way to calibrate our ears. After spending some time on our mix our ears will get used to it and we will lose perspective. By listening to reference tracks, we can re-establish a sense of context, of what a conventional piece of music sounds like and how our piece may sound next to another in a listener's playlist. Whether we then want to try to make our mix closer to that of the reference, or even exaggerate their differences is entirely up to ourselves. What matters is just that we don't get lost in a vacuum after hours of working on our mix.

Another advantage of reference tracks is that they can help is quickly get used to different listening environments. When you switch to your headphones and suddenly your mix sounds incredibly boomy, what are you to make of that. How do you know whether it's the headphones or your mixes that are weird. Well, you listen to some references on those headphones first. If they are boomy too, then it's likely just the headphone's fault and you need to just take a little to get used to them.
Is your mix uniquely off sounding on the headphones? Then there might be an issue with your frequency balance you need to address.

18. Different sections need differerent processing

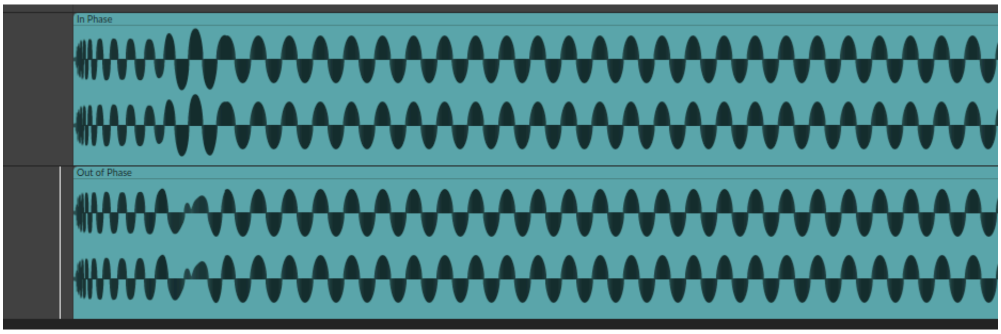
The processing each instrument needs depends heavily on the context around it. As the context around an instrument changes, so will the things you need to do to it, the way it wants to be EQ'd, compressed, etc.
So if you have a lead sound that is playing both during the drop of your song, where the mix is full of drums, basses, pads, etc, and during a breakdown, where it only needs to share its space with some ambiences, create two versions of that lead sound.
Optimise it for each context separately.

You can archieve this with automation, but sometimes the easiest thing to do is just to copy the track it sits on and create two different versions for the two different contexts.

19. Phase Alignment

The phase relationship between two sounds can matter whenever they are very very similar in their waveform. For the most part this only is the case for two recordings from two different microphones of exactly the same event.
However, there is a special case in electronic music. The tail of a kick drum and a sub bass have two incredibly similar waveforms, they are both essentially sine waves. Phase relationships between the two can change the impact of your low end.

During the start of the kick the sub is usually sidechained away, but during its tail end, they tend to briefly overlap.
In this example we can see one render where the kick is exactly in and one where it is exactly out of phase. Notice the dip in volume and the less smooth transition between the two different waveforms in the out of phase example.



The difference between sound is subtle. It becomes more noticeable the louder your playback volume & the more sub heavy your system is.
Whether you go through the tedious process of aligning every single kick with your sub bass is up to you. With more complex basslines and drum patters often a static alignment is not sufficient, you will need to automate the polarity of each kick drum.

The compromise I tend to go for is to make sure the two are aligned on the start of every bar, so I get maximum impact there.

20. Sidechain after Mastering

An interesting observation I've made is that after setting up my mastering limiters, I will often want to go back in and readjust my sidechaining. Because the limiter squashes all the stems together, I find myself increasing the depth and length of my sidechain ducking to kick and snare. What sounded excessive before limiting now sounds nice and clean, whereas what sounded sufficient before limiting now seems lacking.

If you can, I would suggest trying to keep your sidechain compressors live and not rendered down during mastering. After setting up your mastering chain, try revisiting them, to see if you make a similar observation.

21. Pre-Delays for Depth Hierarchies

You may be familiar with the idea that a reverb's pre-delay has a relationship on the feeling of space created by it. Because the speed of sound isn't that fast, in acoustics delay is equivalent to distance. More pre-delay creates distance between the dry and the wet signal, making it sound like the dry signal is closer to you, while the reverb happens further down the hall. You can use this phenomenon to create hierarchies of depth, by sending different sounds to the same reverb, but not directly, but using different amounts of pre-delay.
Instead of sending sounds to the room reverb directly, in this screenshot I send them to intermediary sends delaying the sounds, which then get forwarded to the reverb.

This way I can e.g. send my drums to the reverb with a higher amount of pre-delay, making them feel closer and more separated from the room, while I can lower the pre-delay on my pads, making them sound like an event happening further down the hall.
Keep in mind that pre-delay alone is not sufficient to create a feeling of distance, many other factors, such as volume dry to wet, relationship and stereo width all contribute. This is just one of many aspects of creating an illusion of depth in your mix.

22. Panning with Delay

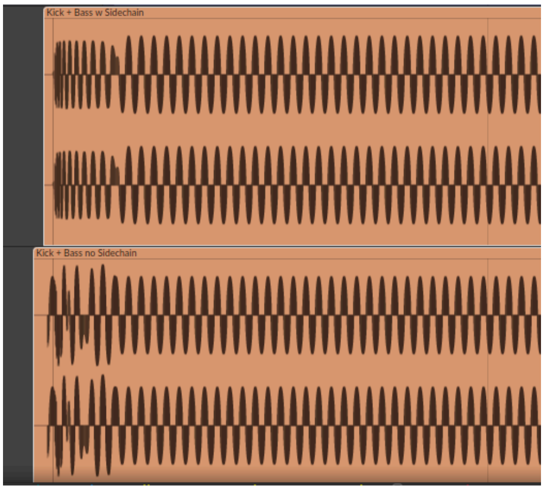
Instead of panning by increasing or decreasing the volume of a sound in the left/right channel as is conventional, you can also pan them by delaying them between either ear.

If a sound arrives later at the left ear than the right, we will take this as a spatial cue, that will make the sound seem like it's coming form the right.
Note that due to the small distance of our ears, we are talking about very small delay times here, you often see the number .7ms, ive personally found that going up to 2ms gets the most convincing effects, whereas higher numbers tend to sound less clearly panned and more like a form of room reflection.

We tend to use these delay cues especially in the lower frequencies. In the top end, the dominating effect is the gain difference between left and right ear, caused by our head blocking the sound. However in the low end, the wavelengths are too large for our head to block them and time difference takes over. This is why panning a bass sound with volume can sound very unnatural, especially on headphones, whereas panning it with delay works fine.

23. Sidechain for Headroom

One of the many reasons we use sidechain ducking is that it wins us back some headroom. When kick and bass add together, their sum will peak a lot higher than if the bass is turned down while the kick is playing. This can help take a lot of load of our master processing like clippers and limiters, without really reducing perceived loudness.



A look at an oscilloscope/waveform can really help us in finding the optimal amount of sidechain ducking to get the most out of this headroom gain without over-ducking the bass.

24. Mono Compatibility

Mono compatibility is the idea that your music should still sound good when it is played back on a single mono speaker. The best way to check for mono compatibility is to simply play back your master in mono and see if the mix still works. It will never sound exactly as good as in stereo, but it should still sound fine.

How much of a priority this takes greatly depends on your intended audience.

If all you really care about is how your music sounds to a headphone listener, mono compatibility is the least of your concerns.
If your mix needs to still sound good on phone and laptop speakers, you need to make sure it doesn't fall apart in mono.
If your priorities are club & hifi playback, where often the low end is played back through a single sub, whereas the high end remains stereo, your focus should be on mono compatibility in the bass range, while you can be more lenient in the higher frequencies.
There are no perfect rights and wrongs here. Ruthlessly optimising for mono compatibility no matter what will limit your creative options, so sacrificing it can be the right call sometimes. It all depends on context.

25. Pitch-Shifting instead of EQ

An alternative way of adding high-end to your sound with EQ is to use pitch-shifters in parallel.

This can have the advantage that it actually shifts information from one frequency range into another, whereas EQ can only turn up what is already there.

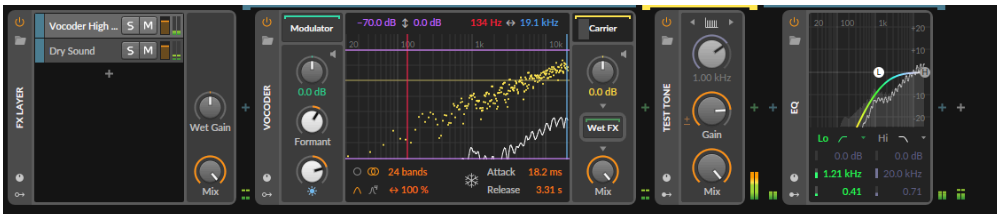
Keep in mind that different pitch-shifters will have very different sounds & artifacts. Pitch shifting algorithms vary a lot, whether it uses autocorrelation, where it preserves formants, whether it works granularly in the time or spectrally in the frequency domain will all lead to different results.
Try out different pitch shifters and see which one works best for the sound & style at hand.

26. Digital Gain Staging

Gain staging in a DAW works a little differently to analog gear. DAWs tend to work in 32 or 64 Bit floating point and come with a lot of headroom, as long as your sounds do not peak on the master, individual tracks having huge overshoots will have little effect. The issues only come when rendering out the master, 24 or 16 bit files both clip at 0dBFS, which can lead to harsh, aliased distortion.
Whether the gain coming into a given effect matters all depends. Nonlinear effects like dynamics processors, clippers or analog modeled filters will all behave differently depending on the incoming sounds volume.
Other effects, like digital EQs and filters are linear, they do not care if you turn up a sound before or after the processor at all, and the results will be exactly the same.

27. Vocoders as Exciters

An interesting, and smooth way to add high end to some sounds, especially percussion, can be using a vocoder modulating some noise. The noise will be made to follow the envelope of our sound, while also adding some new information, especially if we mess with attack and release time. We can then highpass filter this noise and add it in parallel to our sound.



Make sure to mess with parameters like band count, bandwidth, formant, brightness, attack and release to get the optimal layer. Which parameters are available to you will of course depend on which vocoder you are using.

28. Keytracking instead of EQ

When we EQ complex synth parts like arpeggios, often we really only need to turn down certain notes inside the part. If we still have the midi and synth running live, an interesting alternative to EQ can be to simply use a keytracker to turn down the volume of our synth at problematic pitches. This way all other notes will be left completely unaffected.

29. When to use the Solo Button

Usually it is best to make mixing decisions in context. However the solo button can really help us focus in on the details of a certain sound. Here's a simple guiding principle you can use when to work on a sound in solo vs while listening to the full mix.

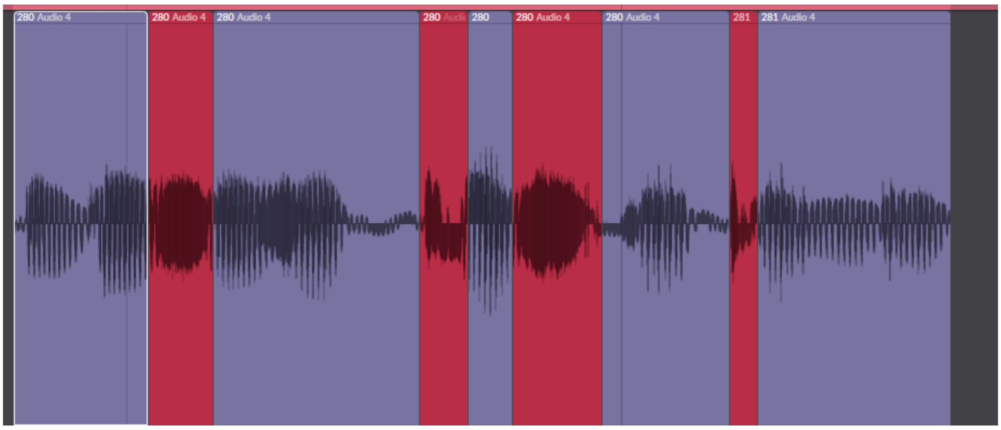
While removing from the sound, listen in solo. You are at no risk at making it take up more space and you might hear some more things you can remove from it that may be masked when listening in context.

While adding to a sound, listen in context. Make sure that what you're adding to the sound isn't getting in the way of other mix elements.

Of course these aren't really hard rules. I certainly don't follow them all the time. It's just a possible guiding principle to use when it helps and abandon when it doesn't.

30. Why you often need to De-Ess after compression

When taking a look at a waveform of human speech, we can broadly categorise it into two components. Pitched low frequency vowel sounds (purple) and noisy, high frequency sybilants (red) like s, t and other sharp consonants.



When compressing a waveform like this, the compressor will mainly react to the lower frequency, higher amplitude sounds, acting less on the siblances. These energy dense high frequency components sound louder to our human ear than to a compressor, so they will often end up being disproportionately loud after compressing.

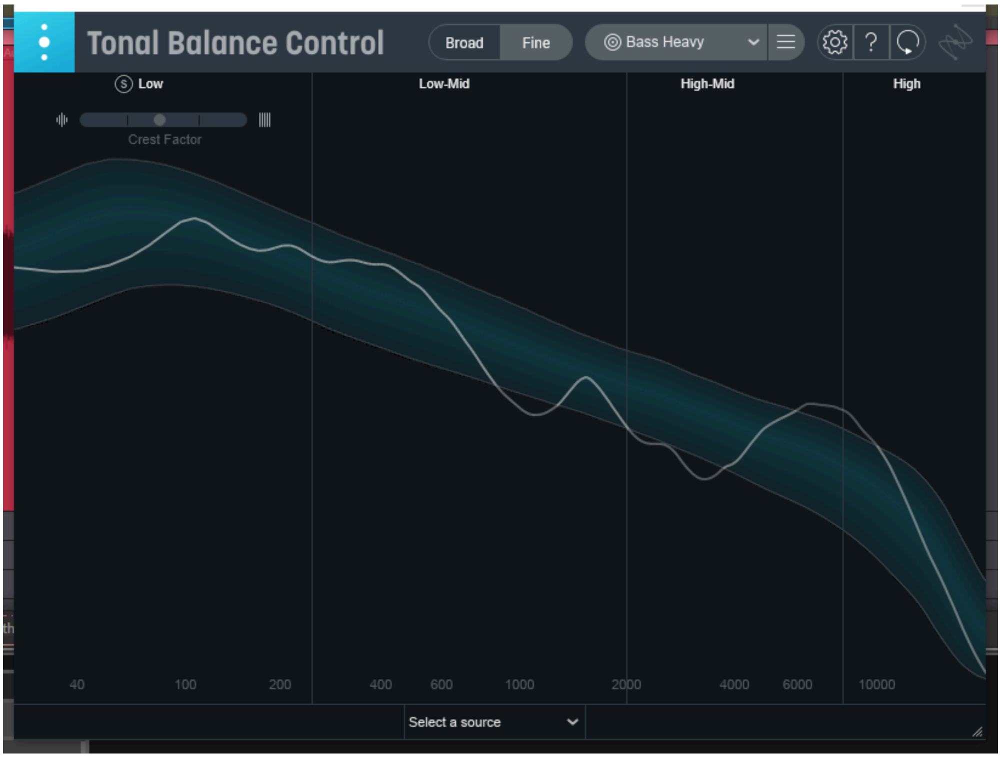
To compensate for this, we use a de-esser. In its simplest form, a de-esser is just a compressor designed to be especially sensitive to the high end. This allows us to control these sharp bursts of high-end noise, that conventional compressors tend to underreact to.
Many different, more sophisticated designs of de-esser exist nowadays, but the basic goal remains the same. Control the high-end that stands out too much after compressing speech.

31. Volume Meters do not show perceived Loudness

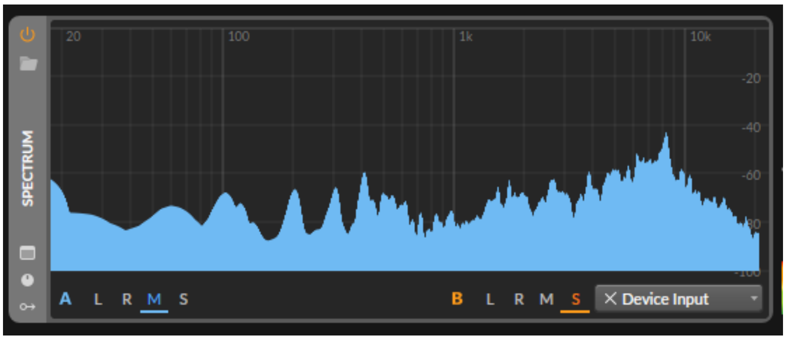
Your DAW's gain meters are not a great tool to get an insight into your mix balance. They only care about amplitude, nothing else. Our ears however hear certain frequencies as much louder than others. A 40hz sine wave and a 2000Hz Sine Wave peaking at 0dBFS will look identical on your meters, however one will sound a lot louder than the other. Use your ears to judge the balance between sounds, not a decibel reading.

32. Using Analyzers where Speakers fail you

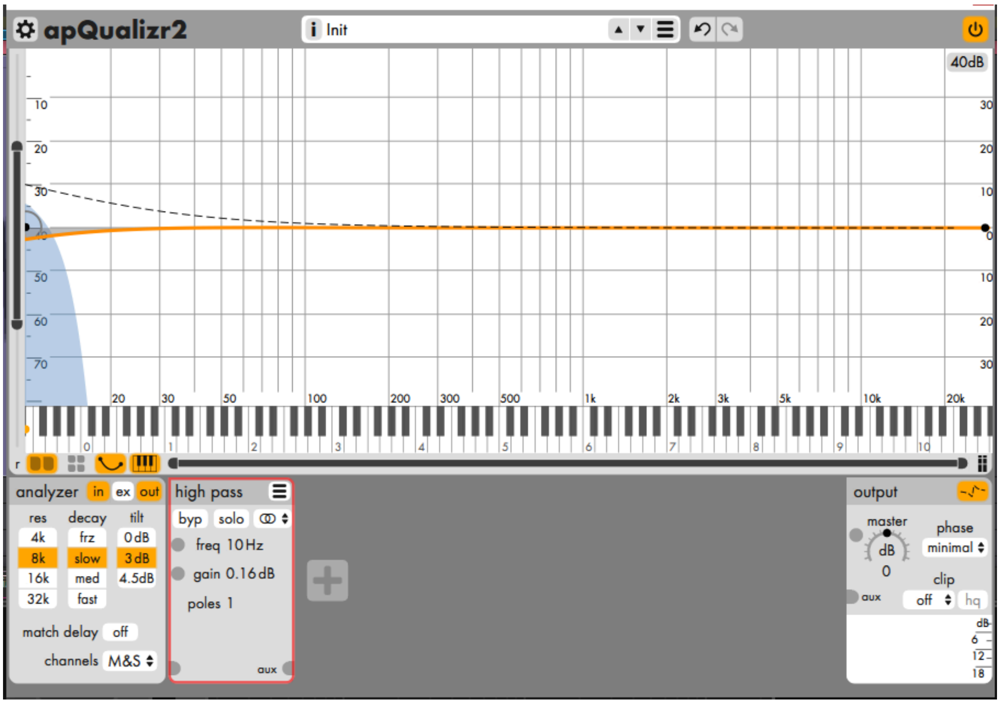
The upside of analyzers is that they can give you objective information about parts of your mix where your ears and monitoring system will fail you. These are usually the extreme top end, due to high frequency hearing loss and the volume of your sub basses, due to room & speaker problems.
Using spectrum analyzers to compare the volume of these areas against some reference tracks can help you overcome or at least compensate for deficiencies in your signal path.



Another problem spectrum analyzers can show you that your ears won't is DC offset, it will show up as some extreme low end, 5hz and below. This is completely inaudible, but can eat up a lot of headroom. It's especially likely to show up if you like to dabble in some more unconventional sound design and processing techniques.

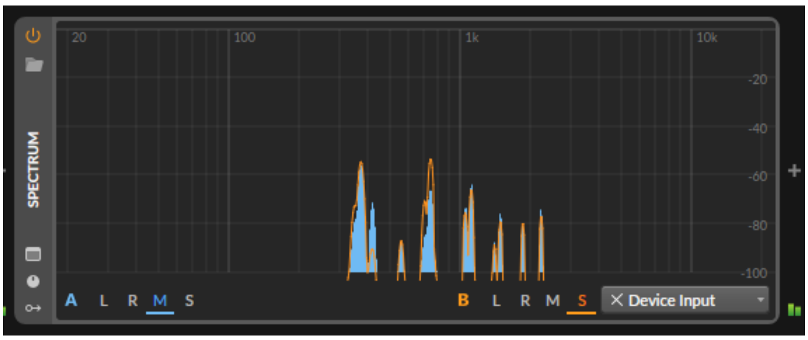


Filtering this off can be done by simply using a very shallow high pass filter as low as your EQ will allow you to set it, e.g. this 6db/O filter at 10Hz in apQualizr 2.



33. Resonances and when they're a problem

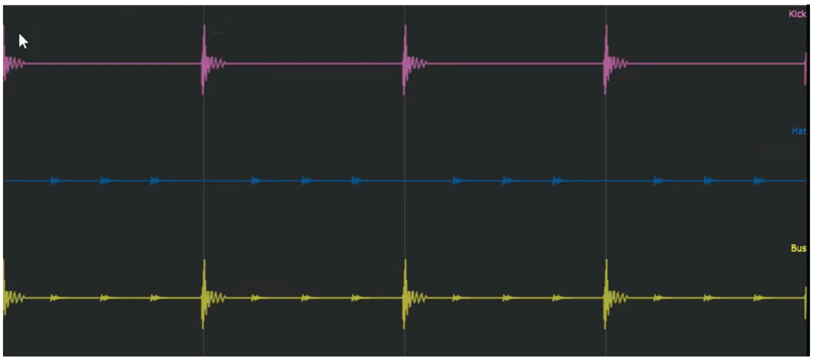
Resonances are commonly talked about as if they are always an issue you need to fix. However, any kind of pitches in your music are created by resonance. Certain kinds of instruments, like flutes, will show up on your spectrum analyzers as pure resonance. These are not problems to be fixed, they are the essence of the music.



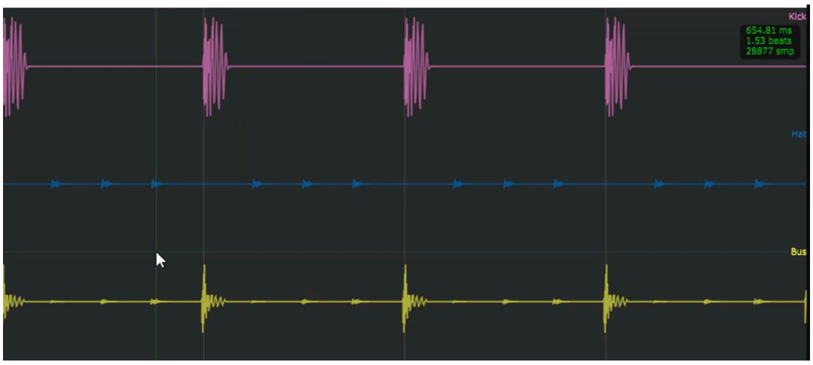
Resonances only become a problem when they're unwanted, because they're out of key or ring out for too long. These will not necessarily be the resonances that show up the loudest on your spectrum analyzer and you cannot trust a resonance removal plugin to be able to tell good from bad resonance. If you want to use a resonance remover, try to listen closely first and make sure that it is actually removing the parts of the sound that are causing a problem and nothing else. Use the EQ and Attack Time to strictly limit the plugin, so it will not remove anything else and flatten your mix as these tools can have a tendency to do.

34. Bus Processing vs Individual Tracks

Bus Processing can do certain things that processing individual tracks cannot. With some linear effects like EQ there will be no real difference, but anything that reacts to a sounds loudness, like compression or saturation will have very different effects whether theyre applied on a bus or on each track separately.
If you e.g. compress a kick and a hi-hat separately, notice that the hi-hat isn't really affected by the heavy compression of the kick.



If we compress them together on a bus, we can see the kick pumping the gain of the hihat due to the release time of the compressor. The hihat is made to react to the kick.



Effects like this are the very essence of mix glue.

35. Sends vs. Processing on a Track

You have maybe encountered the notion that the professional way to apply reverb to a sound is to do so in parallel at 100% wet by placing the reverb on a send instead of using the dry/wet of the reverb on the track. However, each approach can have its pros and cons. Here are some things I think about when choosing which way to go.

On a Send

- saves CPU (multiple tracks can use the same reverb)
- creates coherence (multiple tracks in the same reverb sound like they're in the same room)

On Each Track

- bespoke processing for each sound
- processing reverb & sound together after mixing them
- reverb as a part of the sound design

36. Mixing into a Compressor

A compressor/limiter/clipper on the master will affect your perception of the mix balance, because it turns loud sounds down more than quiet ones. If you know you will be placing this effect on the master at some point as a stylistic choice, it can make sense to mix into it from the start.
This allows you to make mix decisions with how the mastering processors will react to them from the start.

I personally tend to mix into a clipper from the start and do the rest of my mastering when the mix is finished. What you will do is up to your personal preference and style.

37. Objective vs. Subjective Aspects of Sound

An error you see a lot of people make when giving mixing advice about the "right" and "wrong" way to do things is conflating the objective and subjective aspects of sound.

Objective truths are ones which can be mathematically demonstrated or scientifically studied. These things can be objectively right or wrong, regardless of taste.
These are things like:
-When the master is clipping, it will experience harmonic distortion & aliasing when rendered to a 24-Bit Wav File
-Using Haas-Delays for stereo widening will cause comb filtering in the mid channel, compromising mono compatibility.
Subjective aspects of sound on the other hand are value judgements ultimately up to taste and artistic vision. This is the A causes B, therefore it is wrong. The"A causes B" step can often be clearly demonstrated, it is not up to debate. "Therefore it is wrong" however is up to opinion.

Consider statements like:
-Clipping and aliasing on a master always sound bad and are therefore simply incorrect
-Mono Compatibility always needs to be a number 1 priority

You can accept the objective part of someones argument, but then reject the latter half. Whether you like a certain sound or not is not something anyone can dictate, no matter their knowledge and authority about the subject.

Keep this distinction in mind, because it is highly likely to matter at some point, especially if you work in more experimental genres of music. It will allow you to still learn from people while disagreeing with them on their final conclusions.

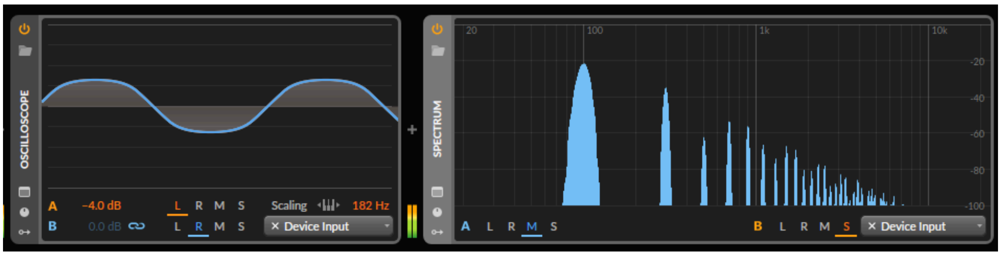
38. Bounce to commit

Sometimes you get stuck on a mix. Often this is caused by an inability to commit. A techniquel I sometimes use to force myself to move forward is to render every track to audio as it currently is when it's time to move on. This removes a lot of options I would otherwise spend endless time tweaking. I can still work on the mix after this of course, but I get a bit of a fresh start.
An even more aggressive step would be to not just consolidate single stems, but actually render multiple tracks down into one stem, such as bouncing all your melodic elements to one track, all your drums to another. Reducing

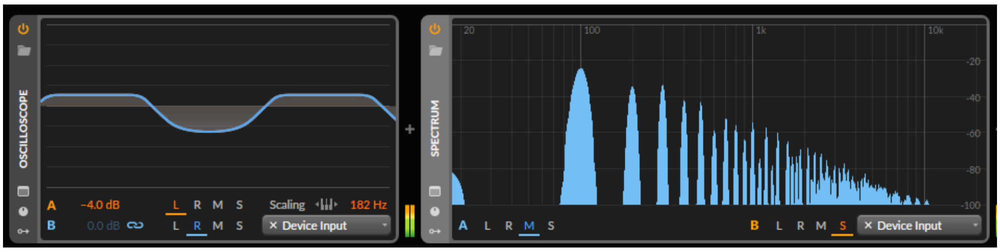
your options to tweak can really help you move forward. Of course you need to make sure that none of your routing breaks when you do this.

39. Odd & Even Harmonics

When saturating a sound, we have a few different options. One of the most important distinctions is whether our saturator only adds odd, or if it also adds even harmonics. Saturators that add even harmonics usually show an asymmetrical transfer function, whereas ones that only add odds treat the top and bottom half of the waveform the same. Tape saturation is often modeled with symmetrical soft clipping, while tube saturation is usually digitally modeled with an asymmetrical transfer function that also creates some even harmonics. You almost never encounter a saturator that only adds even harmonics, when people say X Saturation adds even harmonics, it is always understood that it also adds some odds.
Here is a simple symmetrical soft clipper adding only odd harmonics to a 100Hz sine wave:



And here is an asymmetrical soft clipper also adding some evens. Notice how it treats the top and bottom of the waveform differently. Asymmetrical waveshaping can often introduce DC-Offset, so it is often sensible to highpass a sound at 20Hz or lower after using asymmetrical saturation:



40. Sub Replacement

Sometimes rather than trying to really control your processing on a bass sound, it can be easier to just highpass out the low end at around 100Hz and layer in a separate sine wave sub with some harmonics. This also allows you to sidechain the sub differently to the mid bass.

By making sure it plays the same notes & has a similar envelope, you can ensure it perceptually fuses into a unit with the mid bass if you wish it to. Subtly sidechain expanding it to the mid bass can be very effective.

41. Tonal Balance Sends

We can use sends with parallel filters on them to adjust our tonal balance.
The idea is as follows:
- Create a send with a bandpass filter for a given frequency range you want to reinforce
- Send any tracks to it you want to use to enhance that frequency range
- use dynamics processing, reverb, saturation,... on the bandpassed parallel signal to taste

When using filters in parallel, make sure to either use linear phase filters or a low pole count.
When using filters with steeper slopes than 6dB/O or sometimes 12dB/O you will cause phase cancellation with the dry signal.

42. Priority Sends for Unmasking

Priority Sends are a type of send track I sometimes use whose sole Purpose is to trigger a sidechain compressor/spectral unmasker. The send does not put out any audio itself, it remains muted.
I send all instruments i want to be in the foreground and clearly audible to this send track.
I then use this send as a sidechain source for plugins like outlined in chapter 9.

The advantage of this is mainly workflow speed and some flexibility, it allows you to use a single plugin to duck a sound reaction to more than one sound.

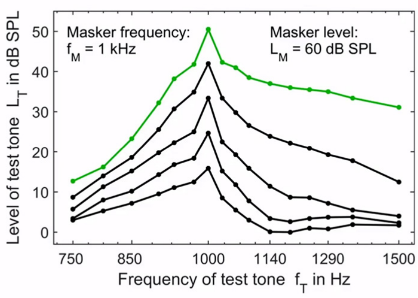
43. Transient Sends

Transients Sends are a way for me to reinforce a sound's transient in parallel.
I send sounds like kick & snare to this send & then use transient shapers & gates to isolate the transients. You can do this using the multi-outs of a plugin like ST1B or simply turn down the sustain value of a plugin like the free kilohearts transient shaper. I then use clipping, limiting & expansion to further sharpen and tighten this parallel transient signal until what I'm left with are pure snappy clicks.
Type B saturation as outlined in chapter 65 is your friend.

You can then blend this transient signal with the dry sound to really reinforce its snappiness.
Doing this on a send allows you to reinforce the transients of multiple sounds at once.
The quality of the transient shapers you use can matter a lot here, try some alternatives.

44. Sounds Mask Upwards

Masking is the phenomenon of one sound overpowering another sound, making it harder to hear. I've made a full video just on the psychoacoustics of this I'd recommend you check out if you've never heard about it.
Sounds generally mask sounds that exist in the same frequency are as themselves. Due to the structure of our inner ear however, sounds are more likely to mask sounds above than sounds below them in the frequency spectrum. This is known as the upwards spread of masking.



If one element in your mix is hard to hear, look not just in its exact frequency range, but also slightly below it. This is probably the reason the low mids get so much hate.

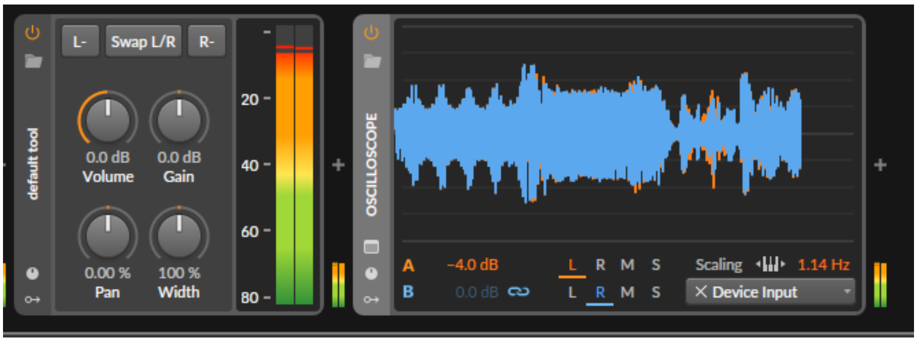
45. Stereo Width = Distance

This is a very simple, but surprisingly effective principle. The header is misleading though, it should really say Stereo Width = 1/(Distance) since a soundsource sounds wider the closer it is.
This makes sense intuitively, a closer object looks wider because it takes more of your field of view & the same

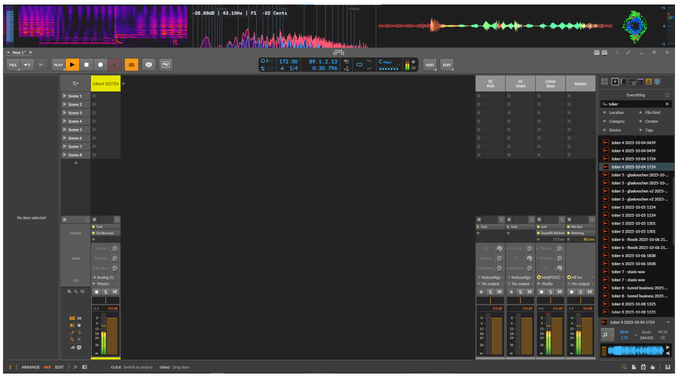
happens for its sound.
Just using a width knob on utility or tool can already get you very far and I was really impressed by how convincing this effect is. It's a great tool to create additional depth in your mixes and place sounds in space.

46. Oscilloscopes vs. VU-Meters

I am a strong proponent of the idea that Oscilloscopes are superior to standard volume meters in every way. They don't just tell you how loud a sound is at any given time, but *when* it was loud & for *how long*. This can help you find the source of overshoots a lot more quickly & estimate much more accurately how a given sound will react to clipping or limiting.



I never look at my DAWs peak meters, I just always have the minimeters waveform running at the top of my screen. It tells me everything peak meters could & more.



47. When to worry about linear phase

The default assumption you should make is that you don't need linear-phase-EQ on a sound & that minimum phase EQ is perfectly fine. Using linear phase all the time will increase your CPU load and latency much more than necessary. There are a few scenarios where linear phase does help however, here are 3:

1. Parallel Processing/Correlated Signals:
   When processing two versions of the same sound in parallel, minimum phase EQ with higher slopes can cause phase cancellation and notch filtering between the two. The same can happen for multi-mic recordings of the exact same audio event.
2. Mid/Side EQ on panned sounds:
   The phase shift caused by steeper minimal phase EQs can cause strange panning artefacts when used in mid/side, which can lead to information from one stereo channel ending up in the other. When highpassing the sides of a panned signal or a master, I tend to reach for linear phase EQ to make sure my stereo imaging remains intact.
3. Clipped Signals
   A clipped sounds peak values rely on the phase of each of its partials. Using minimum phase-eq on a sound after clipping it can easily negate the gained headroom, because the partials' phase relationships change.

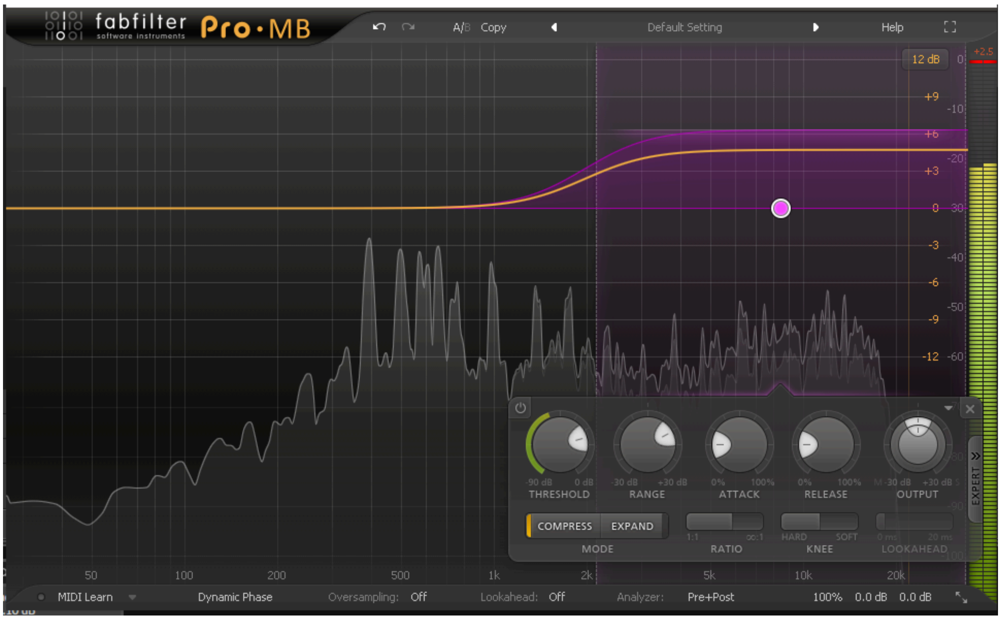
Linear Phase will not always keep the headroom perfectly intact, but it will generally preserve it better. Take note of the oscilloscope and waveforms in these two examples, the only thing that changed is minimum vs linear phase filtering.

I personally try to avoid using linear phase unless I have a really good reason to favor it. It also just sounds different, but those differences are quite subtle.

48. Multiband Dynamics as Gain-Specific EQ

One way to think about multiband dynamics is to frame it as a volume specific EQ. Instead of just boosting the highs, you can choose to only boost them when they are loud or quiet. Let me just give you some examples:
Upwards expanding the highs, only boosting them when they are loud:

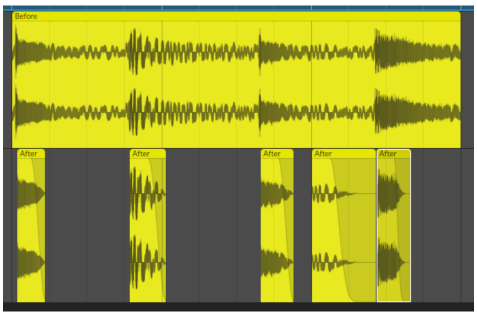
Upwards compressing the highs, only boosting them when they are quiet:



Of course there are a lot more nuances to multiband dynamics processing, but sometimes all you need is a simplified approach like this.

49. The Coke Ears Method

I think it was either Matt Davis or Sean/Noizbleed of Hacienda Audio fame that coined this term. The idea is very simple. Aggresively use clip edits and volume automation to remove every bit of unnecessary information from every track in your mix. This can really clean things up.



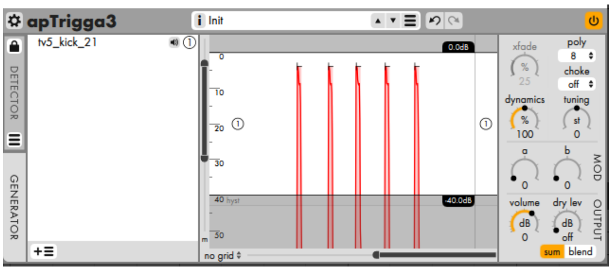
Of course this has a sound that won't be for everyone, but I suggest you give it a try

50. Layering for Frequency Balance

As discussed earlier, EQ can only really add to a sound what is already there. Sometimes if your main mix element needs more high end, what it really needs is not a high-shelf EQ but simply a bright layer of another sound that can provide the necessary high end. Remember to filter this layer so it only adds where you need it to.

51. Drum Triggers & Replacers

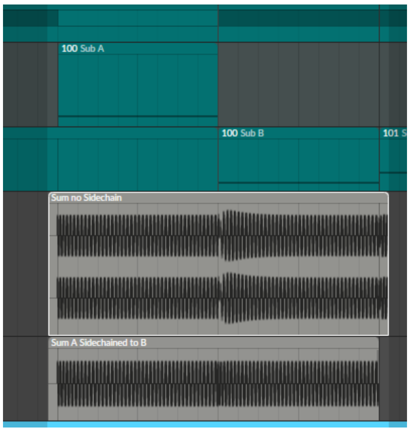
Drum Triggers are a category of plugin that will help you automatically replace a drum sound based on an audio signal. As discussed in earlier chapters, sometimes this is a much cleaner solution than trying to shape that sound to fit the mix with processing. There are many of these out there, your DAW might even have a stock option you can use.



52. Broadband Sidechaining Subs to eachother

Sometimes the tail of one sub bass can overlap that of another creating strange peaks in the summed signal, leading to clipping or other problems.

A simple fix can be to just sidechain one sub to the other, to make sure that when Sub B is playing Sub A is fully quiet.



53. Automated Reverb Tails

The best way I've found to use huge reverb tails without completely cluttering up the mix is to be very selective *what* and *when* you send to them.
Do not have any sounds constantly entering the reverb, only ever automate the sends up for special moments in the composition that really need it.

54. Ducking Reverb

Especially when adding bigger, louder reverbs, they can take up a lot of space in the mix. A way to regain some of that space is by sidechain compressing the reverb, either using the pre-reverb send signal as a compression source, or specific focal elements of your mix like the vocals. This can be especially effective if you use spectral compression like discussed in chapter 9.

Sometimes this will allow you to get the best of both worlds, a really big reverb sound without an overly cluttered mix.

55. Two parts of Reverb

Almost every algorithmic reverb is made up of two components, early and late reflections.

Early reflections model the first bounces off the walls of a room. They tend to be more distinguishable individual delays. The spacing of these delays is how we estimate room size.
When increasing the size parameter of early reflections, you are essentially adjusting the delay times here.

The late reflections or tail of a reverb models sound after it's been bouncing around for a while and has been diffused.
Often this part will offer controls over the frequency response and movement of the reverb. Modulation makes the reverb sound less static, emulating moving air inside a room that resembles a chorusing effect. Bass and high-frequency multipliers let you adjust the decay time for certain frequency ranges. Usually the low frequencies ring out a lot longer, while the highs are absorbed first.

If you learn to make sense of these two parts of a reverb, you will recognise them in a lot of different plugins & be able to use them more accurately and effectively.

56. Spectral Pre-Processing for Reverbs

An interesting technique i've been using to make my reverbs less noisy and overbearing is to pre-process the signal entering the reverb using spectral gates and de-noisers.
This lets me choose which parts of the signal exactly get to receive a long reverb tail. Usually i'm mostly interested in the tonal, pitched parts of the signal, while I don't really need transients and noise in my reverb.

This allows for bigger reverbs that take up less space in a mix. This is obviously not realistic, real rooms do not make such distinctions. It is a stylized approach to reverb not appropriate to all genres of music.

57. Arpeggios as mixing tools

Big pads can take up a lot of room in a mix. If we simply arpeggiate the sound, we still get the feeling of a chord, but with a much more sparse signal. It's a simple but effective way to make a harmonic part take up less room in an arrangement.

58. Adding Resonances

We often talk about bad resonances. However resonances have a really big advantage, they cut through the mix like nothing else. Deliberately adding them to sounds like cymbals can be a great way to make them stand out more.

Of course you do need to listen closely and experiment. If used in the wrong place this technique can create some of the ugliest most unpleasant timbres you will ever hear. In other scenarios it can be a great improvement. I trust you to not use it indiscriminately.

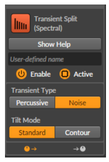
59. Sustain/Transient EQ

By splitting the signal into transient and sustained part before eq-ing, we can gain some finer control over it. This is possible with many different tools, like Eventides SplitEQ or the free Ozone 11 Equalizer.

I tend to just use the transient splits of ST1B together with my stock eq.

60. Tonal/Noise EQ

This one might be a bit of a Bitwig exclusive. Using the spectral device “Transient Split” in noise mode (set in the inspector), we can eq the noisy and pitched components of our sound separately. It’s a very useful technique at times, for which i unfortunately do not know a good way to reproduce it with plugins. You might be able to recreate it using Meldas MXXX.



61. Transient Shaping with Compressors

This is a classic, we can use compressors to boost the transients of our sounds. The trick here is to make the compressor specifically miss the transients and then add makeup gain as needed.
We first compress a sound aggressively, with 0 attack & release and a high ratio.

We then increase the attack time, to make the compressor compress *anything but* the transients. This will make them stand out quite a bit.

This is actually closely related to how a lot of transient shapers work behind the scenes. I go more in depth on this in this chapter’s youtube episode.

62. Keytracked Crossovers

When mixing a mid bass with a sub bass, we want to remove the sub-range of our mid bass to make room for the sub we are adding and avoid any conflicts. By keytracking the highpass filter on our mid-range bass sound to its MIDI, we can consistendly remove the lowest few harmonics from it, no matter which notes it plays:

63. Transient Management

Transients demand our attention. This is why we want to control them. Overly loud transients in background elements can make a mix quite grating to listen to and should be controlled with limiters or clippers, to make sure they don't distract the listener from the focal points of our mix.
This is especially important if we work with a lot of experimental and glitch sounds & a good way to make music in those styles a bit easier on the ears.

With each sound ask yourself, do I want this to call as much attention to itself as it does? Or should I tame it?

64. Layers or Hocket

There are two fundamental relationships two mix elements can have to eachother in an arrangement. Understanding how two parts are meant to relate to each other and reinforcing this relationship can help make your mixes a lot more clear:
Layers:
Two sounds are meant to be perceived as parts of one, think different pad layers, snare drum and clap, ...
To reinforce this fusing of the sounds, you can use a lot of techniques, like making their envelopes more similar to eachother, syncing up their start/stop points, bus compression/saturation, a common reverb etc.

Hocket:
Two sounds are meant to be fully separate from eachother. They are meant to contrast against eachother.
To reinforce this relationship, make sure they don't overlap. Sidechain them to eachother, eq them out of eachothers way, cut their tails, ...

Of course breaking everything down into these two categories is an unrealistic simplificiation. However, asking yourself questions like this will help understand the musical intent of an arrangement and make sure you are enhancing, not dampening it.

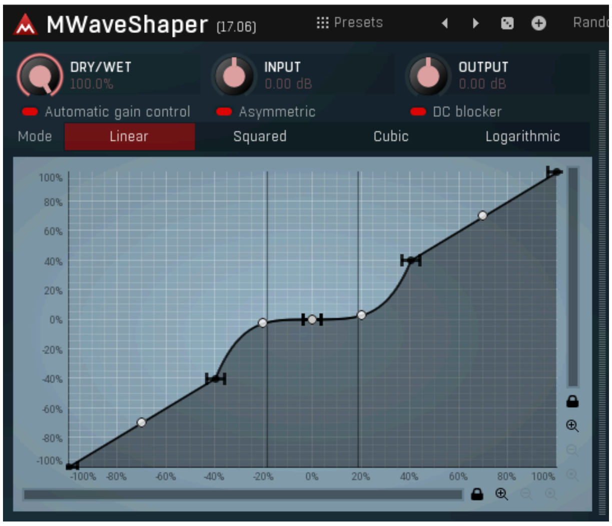
65. Type B Saturation

Type B Saturation is a kind of waveshaping that operates on the quietest part of the waveform.

It also adds harmonics, but has a very different effect on dynamics compared to typical soft clipping saturation. Where soft clipping compresses/limits, this expands/gates, it can enhance a sounds snappiness which makes it very well suited to drums and transient heavy material. You can think of it as audio-rate gating.
This is traditional clipping:

This is type B Saturation:

To create this effect with a simple waveshaper, just draw in a gate-like transfer function like this:



66. Sub-Mixes

A large mix can be overwhelming. To break down complex mixes, it can help to select a certain subset of tracks and take care of any relationships within that sub-group first.
E.g. I often start with just Kick, Snare & Sub Bass solo'd. If elements within a sub-group conflict with eachother, they will also be at odds when listening to a full mix.
Tackling a few sub-groups when the full mix is overwhelming can help create some clarity. Sub-Groups I often work on are:
- Kick, Snare & Sub-Bass

- All Drums
- All Basses
- All Melodic Instruments

67. Frequency Shifting Hi-hats

Frequency shifting a sound up is a good way to add high-end to it without using EQ. While pitch shifting is better suited to tonal instruments, because it preserves harmonic relationships, frequency shifting works best on percussion, because it better preserves timing and dynamics.

68. Modulation instead of Compression

All dynamics processors are just ways to modulate the volume of a sound. If we still have MIDI available to us, an alternative to dynamics processing can be to use midi triggered envelopes to modulate our volume. Instead of gating a sound, we can just modulate its volume with a short release time later down the chain. Instead of compressing it, we can just add more attack time to fade off the transients.

Outro

Trying out all the techniques in this book should expand your mixing tool kit quite a bit.
Just reading about them will be useless, you need to recreate them in your DAW to really remember and be able to use them.

I cannot tell you when to choose which technique. There is no real right or wrong in music.

If you understand them all well and develop an intuition for how they sound, your tastes will dictate your own system of rules for their use.
Only practice and experimentation can get you there.

Thanks a lot for your support.

If you're interested in seeing how I use them in practice & choose between them, my mixing course on udemy shows me mixing 3 songs from start to finish, using many of the tricks discussed in this pdf:
https://www.udemy.com/course/mixing-mastering-electronic-music/?referralCode=D92E3AD4303373CCB6B6
Of course you may disagree with some of the choices I make. I hope you do. That's a very good sign that you're starting to find your own voice in mixing.

If you're interested in taking private lessons with me, head over to https://calendly.com/tildesounds/lessons

For a selection of projects I've worked on, consult https://tildesounds.carrd.co/

You can reach me at tildesounds@gmail.com or on discord under the username tildesounds.

Cheers,

yona