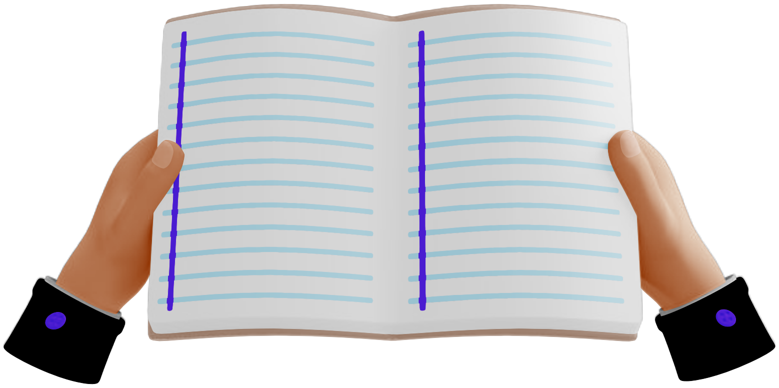
MIXING. SIMPLIFIED.

thomas brett mixing



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INTRODUCTION

Mixing is pretty complicated...

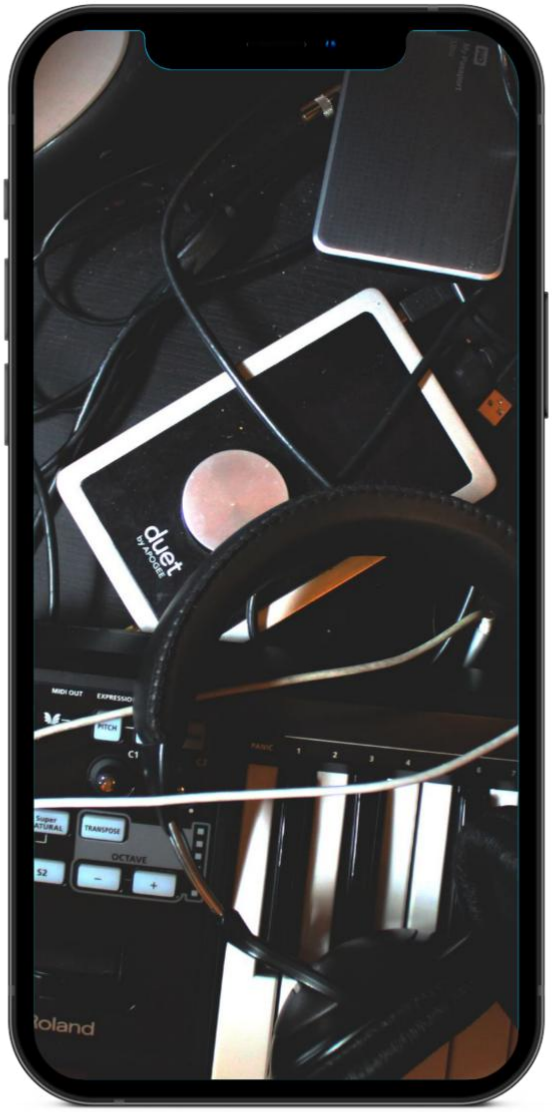
As someone who's been doing it daily for over 15 years at this point, with over 250M+ streams across my portfolio, it's easy to forget that some of the HUNDREDS of tools and concepts which seem *so* straightforward to me nowadays *weren't* always that simple...

Heck! I can't even put into words (or numbers) how long it must've taken me to FINALLY get my head around Compression...

ANYWAY...

Hopefully this guide will help you bypass some of the hardships I went through, and kickstart your journey towards producing professional-sounding mixes.

This, is Mixing Simplified.



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EQ.

SIMPLIFIED.

EQ, or Equalisation is the primary tool of a mixing engineer.

It allows us to *boost* or *cut* certain frequencies (or frequency ranges) to accentuate or lessen certain tonal qualities within a sound.

Using EQ, we can enhance the high-end "smack" or "attack" of a drum, reduce the "nasal" quality of a vocal, "thicken" the low-end of a bass guitar, or long story short, "sculpt" a sound to sound more like what we hear in our heads.

Since no two instruments, voices or microphones sound identical to each other, there's unfortunately no such thing as a "right" EQ setting/curve that will work perfectly 100% of the time.

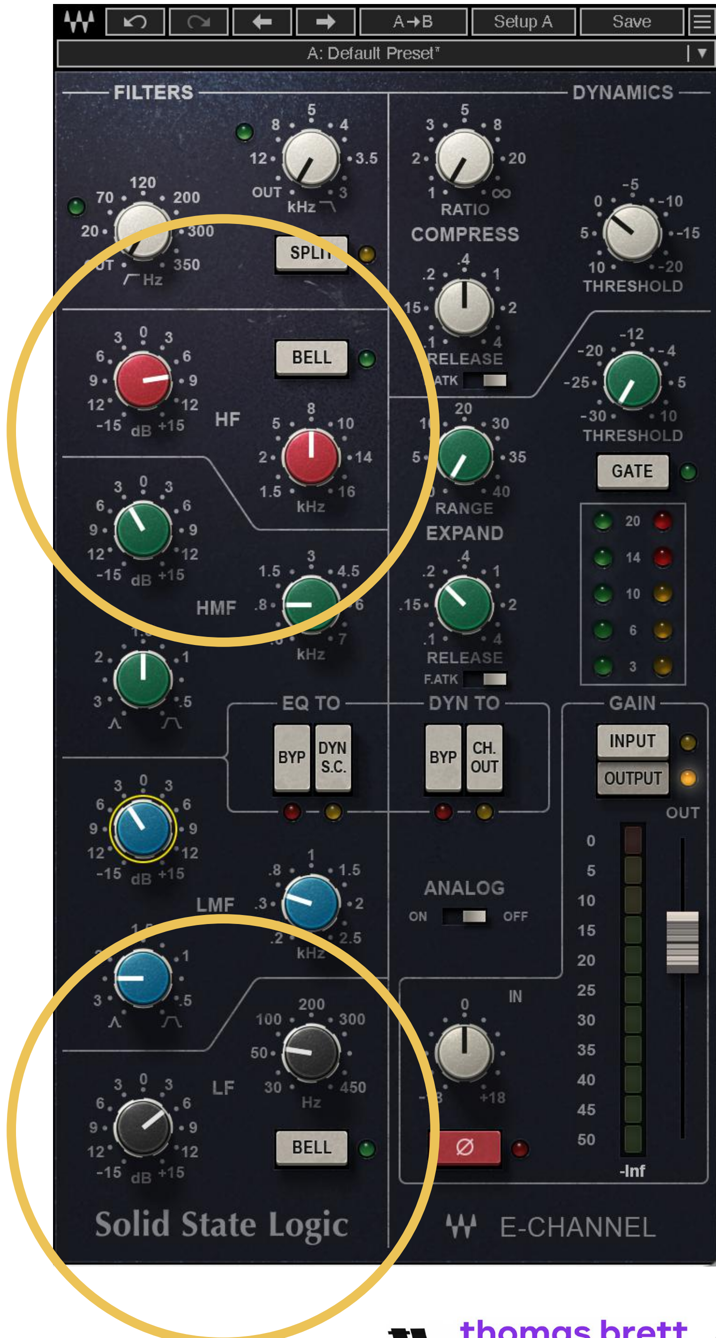
That being said, I do know some *magic* go-to frequencies that will get you most of the way there...

EQ. SIMPLIFIED: RAW SOUNDS ARE FLAT!

Generally speaking, *most* "raw" sounds captured with a microphone will sound pretty dark, lifeless, and "flat" when compared to the "polished" sounds you'll hear on the radio.

While the exact frequencies you choose will shift *slightly* for each instrument, boosting a healthy dose of highs (treble) and lows (bass) with a sort of "happy face" EQ curve is the simplest, quickest way to make pretty much ANY recording sound 1000% better.

Let's take a look at some of my go-to frequency choices for this purpose...

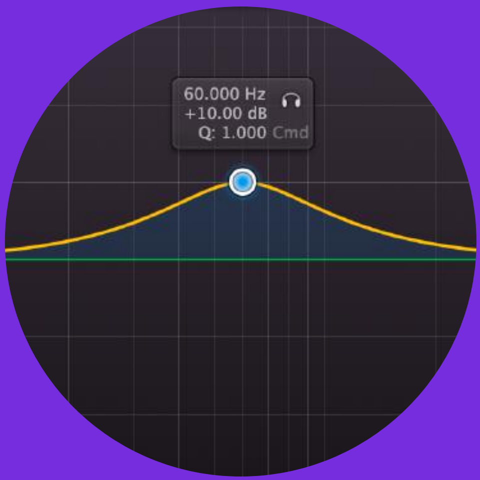


EQ. SIMPLIFIED: "GO-TO" HIGH FREQUENCIES

HIGH SHELF	HIGH (TREBLE) FREQUENCIES TO BOOST
8kHz	If in doubt, an 8kHz SSL-style high-shelf boost is a foolproof way of making most sounds and instruments sound clearer, more "expensive", and *just better* overall 99% of the time.
5kHz	While an 8kHz boost will almost *always* work on pretty much any source, if you need a little more "presence", try shifting the high-shelf down to 5kHz instead!
1.5kHz	Lower instruments like bass guitars, certain synths, or 808s physically *don't* have any useful frequency content above 5kHz. In these cases, try a 1.5kHz high-shelf boost instead.

Q - "But how much do I boost?!?"

A - Quickly boost and then cut +/-10dB to acclimatize your ears to what "bright" and "dark" sound like. A +5-8dB boost will work well most of the time, but don't be afraid to go beyond that if necessary!



EQ. SIMPLIFIED:

"GO-TO" LOW FREQUENCIES

LOW BELL	LOW (BASS) FREQUENCIES TO BOOST
60-80Hz	If in doubt, a 60-80Hz SSL-style bell (or shelf) boost is a foolproof way of making kicks, toms and certain bass instruments sound thick & powerful.
80-150Hz	If you're going to boost low-end on an instrument, you'll usually want to do it around the "fundamental" frequency of the note(s) being played. For bass & guitars, this is often between 80-150Hz.
150-250Hz	The fundamentals of higher-pitched instruments like snare drums, female vocals, and various "piccolo" instruments may live more in the "low-mid" range, and can be boosted between 150-250Hz.

Q - "How do I find the fundamental frequency of a sound?"

A - Use an EQ with a spectrum analyser to visually pinpoint the fundamental bass notes of a sound. Then, use the same cut/boost technique we discussed previously to acclimatize your ears to what thick/thin sound like.

EQ. CONCLUSION.

Beyond boosting treble and bass to make something sound brighter, clearer, thicker and punchier, you'll also want to listen out for anything unpleasant that you may want to *cut* (or reduce) in order to free up some space and get things sounding even clearer.

THE HARD PART...

While it's MUCH harder to give frequency recommendations for cutting (as most problems are specific to the gear, sound, and equipment used during recording), the areas on the left are a few good places to start.

Ideally, the goal is to develop your ears to the point that you can hear, pinpoint, and diagnose problems yourself without blindly cutting to resolve "potential" problems.

COMPRESSION.
SIMPLIFIED.

Compression is easy to use, but difficult to master...

The primary use for "Compression" (or "Gain Reduction"), is to *compress* the dynamic range of a sound or performance, effectively "leveling it out".

With something like a vocal recording for instance, there may be significant jumps in volume throughout the performance, causing it to sound too *quiet* in certain sections, and too *loud* in others.

By *compressing* the louder parts so that they're more in line with the rest of the signal, we can achieve a more *consistent* result which will sit in a track much more comfortably, without sticking out or getting lost.

Simple enough. Right..?

Well... This is where things start getting complicated...

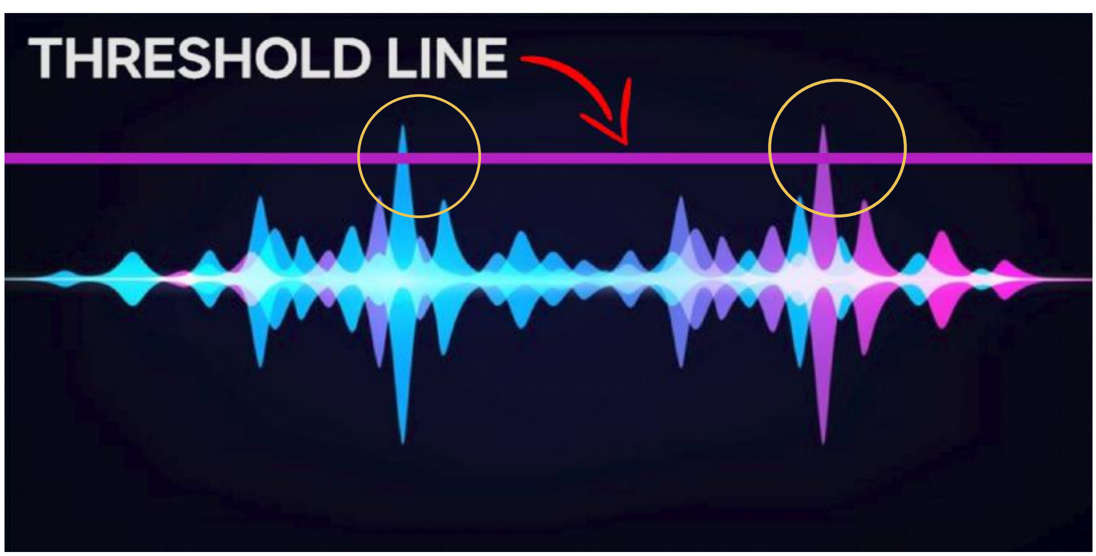
COMPRESSION. SIMPLIFIED:

THRESHOLD & RATIO

THRESHOLD:

In the simplest terms possible: The "Threshold" control on a compressor determines the *minimum volume* at which compression will start to be applied.

Anything which goes above the "threshold line" will be compressed, or reduced in volume according to the "Ratio" of the compressor...



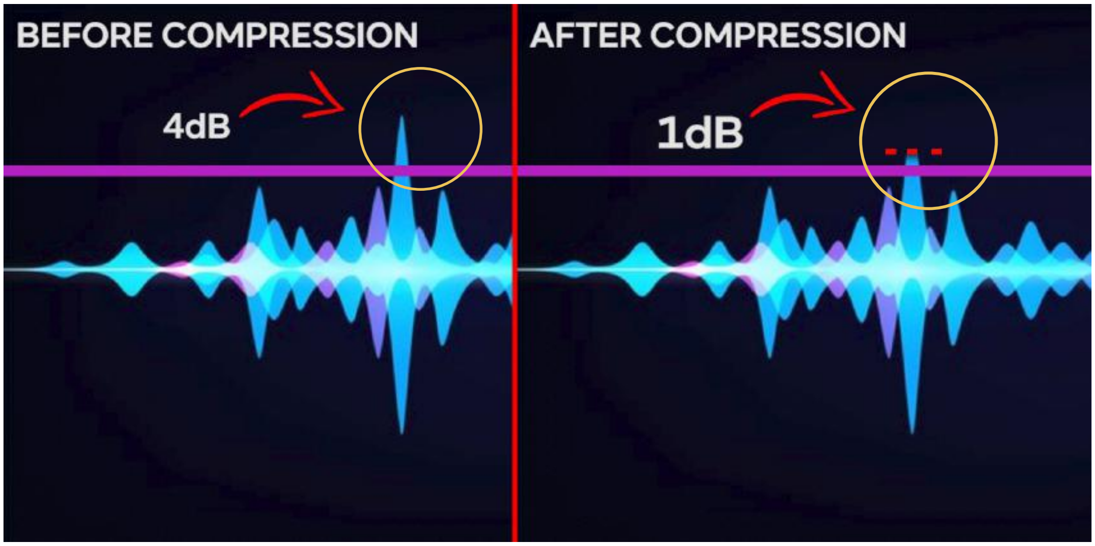
COMPRESSION. SIMPLIFIED:

THRESHOLD & RATIO

RATIO:

At a compressor "Ratio" setting of 4:1, for example, any signal/peak/volume which goes *above (louder than)* our "threshold line" will be compressed, or turned down at a 4dB to 1dB ratio.

So... If let's say, a snare drum peak goes 4dB *over* our threshold, only 1dB of volume will be allowed through (-3dB of compression/gain-reduction will be applied). If it goes 8dB over the threshold, only 2dB will be allowed through...



EASY-MODE COMPRESSION?

Ok. So that's the *easy* part of compression sorted...

Generally speaking, the concept of how the Threshold and Ratio work together is simple enough for most people to get their heads around.

We'll take a look at what the *slightly* more complicated "Attack" and "Release" controls do in a second, but first, I just want to assure you that you can *absolutely* start using compression in your mixes without *fully* understanding everything beyond this point.

"How?" you ask..?

With some super-simple "one-knob" compressors, of course!

EASY-MODE COMPRESSION?

While manually finding the "right" attack & release settings for your particular instrument, sound, context and sonic goal is definitely the "more professional" way of going about setting up a compressor, in a lot of cases, certain "go-to", "classic" compressors & settings will do the job just fine.

Luckily, two of the most beloved and widely used options of *all time* also just happen to be some of the easiest to use!

Meet the 1176 and the LA-2A...

EASY-MODE COMPRESSION?

Both of these compressors have been *top* choices for *top* engineers all over the world for decades.

The black one, the 1176, **is highly regarded as one of the *best* compressors out there.**

It's *fast*, punchy and gritty, and works perfectly for controlling (or pummeling) vocals, drums, guitars... You name it!

Generally speaking, the 1176 is a great compressor choice on *faster material,* and the attack & release settings shown in the image above (attack at 3, release at 7), with a 4:1 ratio will work *perfectly* on 90% of the stuff you throw at it.

After tweaking those controls, simply crank the *input* (threshold) knob to where you're getting the desired amount of compression (5-15dB of gain reduction on the meter), then finally, adjust the *output* (make up gain) knob to match the *pre* and *post-compression* volume levels for accurate before/after comparisons. Simple!

EASY-MODE

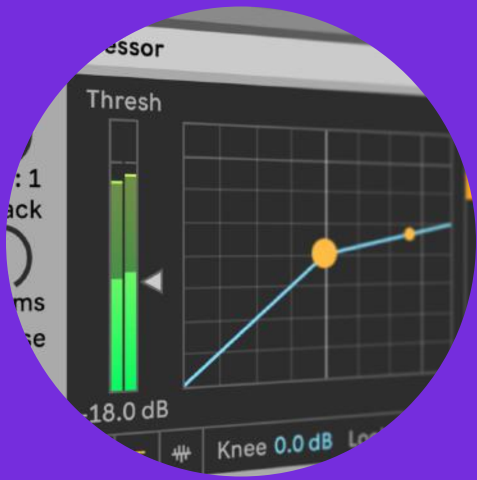
COMPRESSION?

The *silver* one, *the LA-2A*, **is the *slower*, "gentler" brother of the 1176.**

With a *slower* (hidden, non-tweakable) average attack-time of 10ms (vs. the ±1ms attack time of the 1176), and a *much* more gentle release curve overall, the LA-2A is more suited for *gentle leveling* than heavy-handed "dynamic-obliteration".

Generally speaking, the LA-2A is a great compressor choice on *slower material* with less "frantic" dynamic variations, and is the perfect choice for slow, "ballad-esque" vocals, and sustained keys, bass or synth pads.

The 2A is even *easier* to set up than the 76 due to it's lack of attack and release knobs. Simply turn the *peak reduction* (threshold) knob to where you're getting the desired amount of control, then adjust the *output* (make up gain) knob to match the *pre* and *post-compression* volume levels for accurate before/after comparisons. SUPER-simple!



COMPRESSION. SIMPLIFIED:

THRESHOLD, RATIO, ATTACK & RELEASE

While the general concept of "Compression = More Consistent" is simple enough for anybody to grasp, there's a HUGE difference between simply *compressing* something, and *compressing something correctly.*

This, is where the dreaded "Threshold", "Ratio", "Attack" & "Release" controls come into play...

Let's see if we can somewhat "de-mystify" these big words, and figure out how we can use them to get the sonic results we're looking for...

COMPRESSION. SIMPLIFIED:

ATTACK & RELEASE

While the primary goal of Mixing. Simplified. is to give you simplified info and techniques which allow you to dive straight into mixing with ease, I feel it's my responsibility to at the very least, also get you *started* on your journey towards understanding how the "Attack" and "Release" controls on a compressor work.

Although the "easy-mode" compression tricks we talked about previously WILL work just-fine most of the time, having a true understanding of how these paramaters work is the key to GUARANTEEING you achieve the *exact* results you're hearing in your head 100% of the time.

Buckle up folks! We're going in...

COMPRESSION. SIMPLIFIED:

ATTACK & RELEASE

In the simplest terms possible, the "Attack" and "Release" controls on a compressor adjust *how quickly* the compressor reacts-to an incoming signal.

ATTACK:

Once an incoming peak/signal has gone *above* our predermined *threshold*, the *Attack* parameter comes into play, and determines *how quickly* the compressor will reach its fully-compressed state.

The *faster* we set the Attack control, the quicker the compressor will grab onto any incoming peaks, causing it to somewhat "kill" (reduce) the transient smack/attack of the sound, resulting in a *softening* effect.

By *slowing down* the Attack of a compressor somewhat, (typically 10-30ms), we're effectively telling it to *wait X number of milliseconds* before clamping down. This allows the transients to pass through unaffected, resulting in a *hardening* or *attack enhancement* effect.

COMPRESSION. SIMPLIFIED:

ATTACK & RELEASE

RELEASE:

SO... Our signal has gone over the *Threshold*, triggered the *Attack* paramater of the compressor, reached a *Compressed State* based on our pre-determined Ratio **setting (after X number of attack milliseconds), and has FINALLY gone back below the threshold. This, is where the "Release" comes into play...**

While the *Attack* control determines how quickly the compressor reaches its fully-compressed state, the *Release* control determines how quickly the compressor will "let go" and return to it's *fully-uncompressed, default state.*

By *slowing-down* a compressor's release time, we're effectively causing the compression effect to *linger for longer*, pushing the uncompressed parts of the signal *further back*, and causing a *less in-your-face* sound.

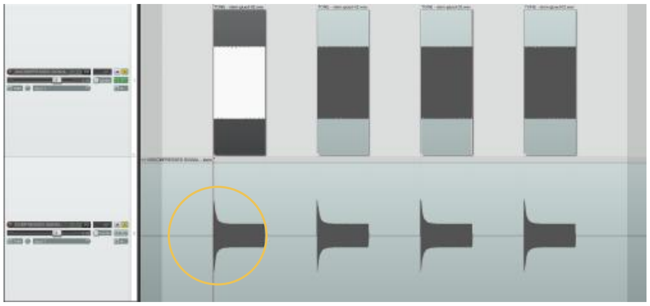
By using a *faster* release time, we're telling the compressor to *quickly stop compressing* when it no longer needs-to, resulting in a *more accurate compression effect*, and *causing an in-your-face* sound.

COMPRESSION.
VISUALISED.

Trying to explain the ins and outs of compression in writing can be difficult...

Personally, I find that it's MUCH easier to understand (and teach) via a few *simple* visual examples:

ATTACK. VISUALISED:



On the left, we can clearly see the difference between the raw, uncompressed 1kHz sine-wave signal on the top channel, and the processed signal below it which has been compressed by -10dB using an LA-2A plugin, which has a ±10ms Attack time.

Due to the fact that it's taking the 2A around 10ms to fully kick-into its compressed state, the side-effect of compressing the signal by -XdB is a +XdB, 10-ms-long increase/boost in front-end transient attack of each note/hit.

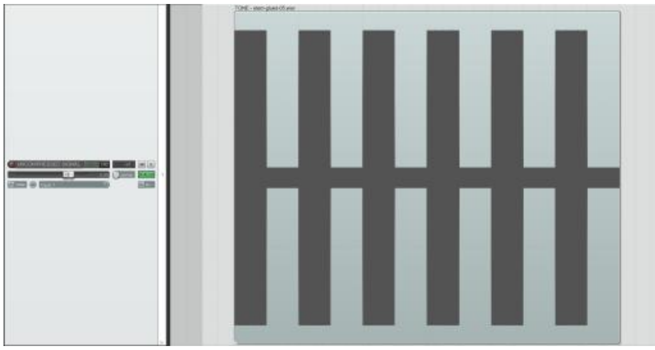
Knowing this, we can essentially use compression as a form of "transient enhancer", alongside the compression we're applying!

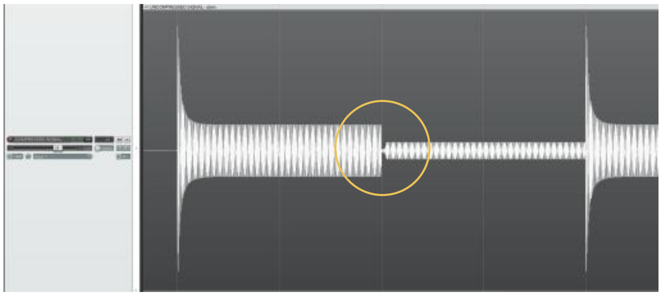
COMPRESSION.

VISUALISED.

RELEASE. VISUALISED:

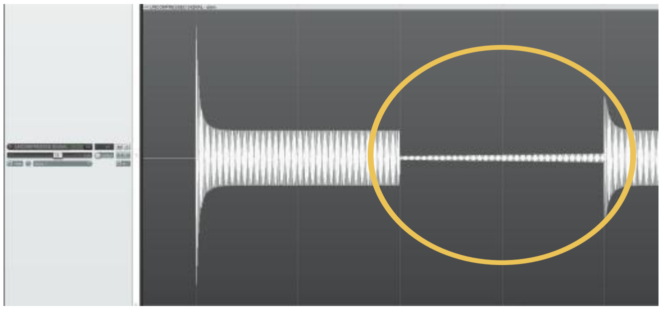
On the right, we have a 1kHz sine wave signal which is alternating between loud parts (which will go *above* our compressor's threshold, and therefore trigger gain reduction), and quiet parts (which w*on't go above* our compressor threshold, and therefore won't trigger any compression).





When compressed using a 10ms Attack-time and SUPER-fast 1ms Release-time, the result is a 10-ms transient boost (as discussed earlier), and a lightning-fast "letting go" or "de-compressing" reaction once the signal has gone below the threshold (marked with the golden circle, on the left).

If instead, we *slow* the release all the way down to 200ms, rather than letting go the moment the signal goes below the threshold, the compressor now takes 200ms to return to it's uncompressed state, even somewhat "eating-into" the attack/transient of the next note! (As you can see marked within the golden circle)



COMPRESSION. CONCLUSION.

Compression is BY FAR the mixing tool which audio newcomers seem to struggle with the most.

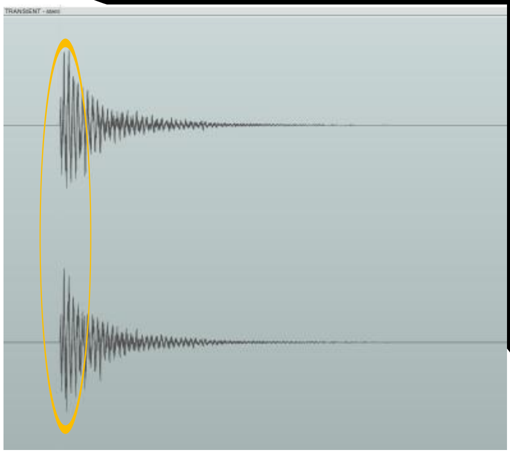
I mean... I can't tell you *how many* long nights I must have spent experimenting with attack and release controls and binge-watching tutorials, just to try and finally get it to "click" in my head...

THE GOOD NEWS IS... IT *DID* CLICK. EVENTUALLY...

Some people prefer to avoid using something they don't fully understand out of fear of failing, but in my experience, just getting on with using it, and learning through those mistakes was the key to cracking the "Compression Code".

So... BETTER GET COMPRESSING!

TRANSIENTS.
SIMPLIFIED.



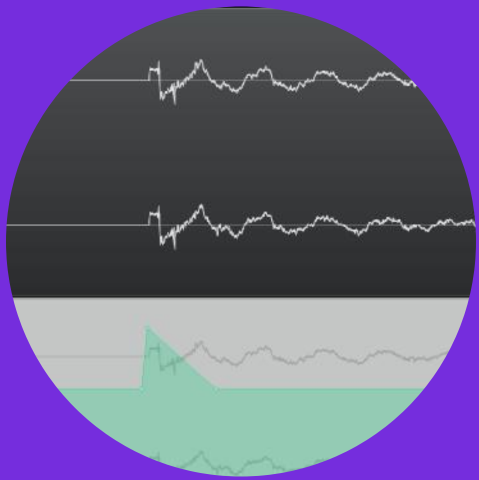
Transients, and how you manipulate them, are *key* to a punchy mix.

A "Transient" is essentially the "Attack", or initial smack/click/pick/etc. portion of a sound.

Transients play a HUGE role in the clarity/definition/overall "Audibility" of a sound, and can mean the difference between one which gets lost in a mix, and one which doesn't.

By accentuating or decreasing the relative volume of a transient vs. the sustain/decay which comes after it, we can *drastically* alter the "Timbre" or tone of a sound/instrument to better match whatever sonic goal we have in mind.

Now. Let's take a look at a few of the ways we can go about doing this...



TRANSIENTS. SIMPLIFIED: MANUAL MANIPULATION

MANUAL TRANSIENT MANIPULATION (VIA VOLUME AUTOMATION):

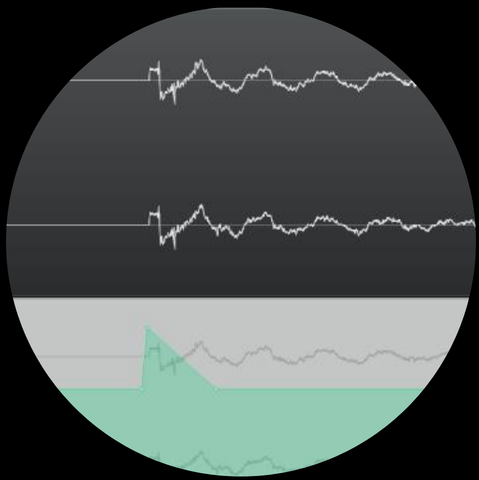
OK. Keeping with our "simplified" theme, let's start by taking a look at the *simplest* way you can go about "Transient Shaping" your audio via *the most* customisable, accurate, and foolproof method imaginable, *without* having to buy any fancy, specialised third-party plugins.

Let's say you're working on a live kick drum recording - It sounds alright, but is lacking bit of high-end "click" or "beater-attack" which would make it hit harder and cut through the track easier.

You don't own any "Transient Shaper" plugins, so you start by boosting a ton of 8kHz high-end with an EQ. Due to the excess of cymbal "bleed" in the recording, you just end up bringing up a ton of noise...

You then try compressing the kick to accentuate the attack, it *kinda* works, but again, also accentuates the unwanted "cymbal wash" in the process...

Back to square one...



TRANSIENTS. SIMPLIFIED: MANUAL MANIPULATION

"Ahh... If only there were a way of boosting *just* the attack of the kick without affecting the rest of the signal..."

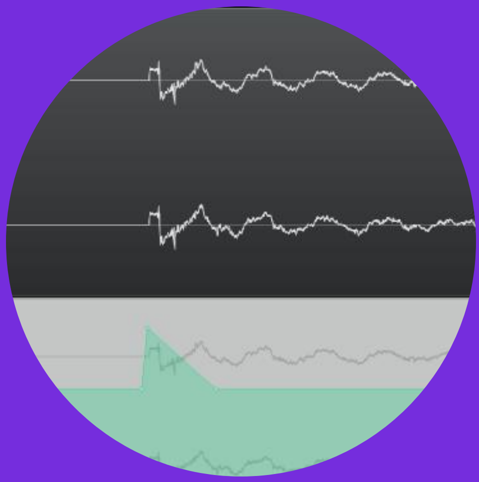
Wait a second...

What if we simply opened up the volume automation lane for the channel, and LITERALLY just turned up the front-end attack of the kicks? (*SHOCK & AWE*)

Seriously though... While it *may* be a little more time consuming than using an "automatic" alternative, doing this task by hand actually allows for a *much* greater degree of control over *exactly* how said transient boost looks and sounds.

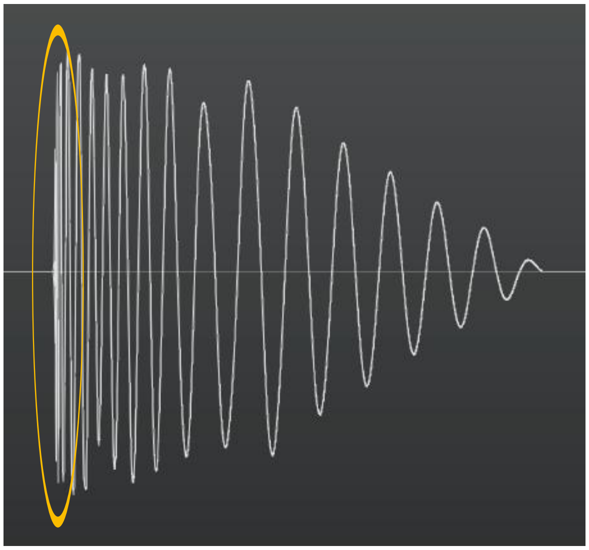
Sometimes I think we've become so obsessed with fancy plugins and techniques that we often skip straight past the most obvious and simple solutions to problems just because we WANT to use them, but I digress...

Let's take a look at a few tips for using this technique!



TRANSIENTS. SIMPLIFIED:

MANUAL MANIPULATION



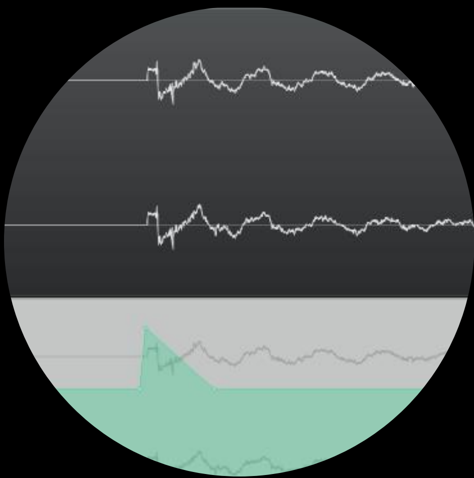
To the right, we have the waveform of a pretty standard electronic kick drum sample.

At the very left (the start) of the sample, you'll notice there's a *tiny* "cluster" of waves which are brighter/more dense than the rest of the signal.

This, is the "Attack" portion of the kick drum, and the reason its "whiter" than everything else, is because it primarily consists of *high frequencies*, which oscillate at a higher rate than lower frequencies, and are therefore more "bunched up" together.

Knowing this, we can *easily* pinpoint the attack of pretty much any sound *visually* within seconds, and then boost/cut/alter it to our heart's content.

Here's how...



TRANSIENTS. SIMPLIFIED:

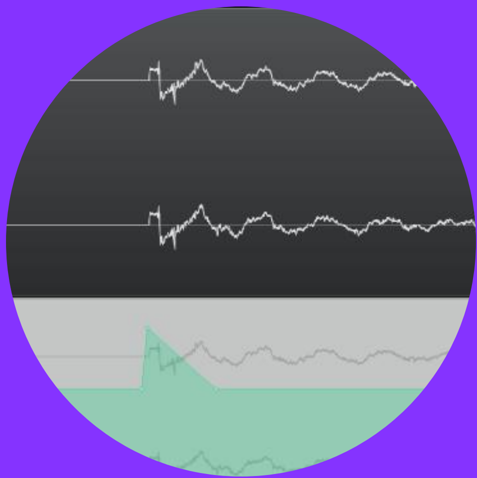
MANUAL MANIPULATION

There are a few different ways you can go about doing this, and none of them are *right* or *wrong*, as it simply comes down to whichever one gives you the results you're after:

WITH CLIP GAIN: Zoom into the start of the waveform. Make a cut/slice where the "Attack" ends (preferably at a "zero crossing point" to prevent clicks/pops, as shown on the left). Boost/cut the volume of the attack using clip-gain as necessary. And finally, insert a small crossover fade where you made the cut for a smooth transition between attack and sustain. Simple!

Granted, this technique works better when using "one-shot" samples which you can simply copy/paste throughout the song with your new clip-gain automation *pre-baked* into each one, as doing this on a per-hit basis with live drums *might* take you a little a while...

As for live drums...



TRANSIENTS. SIMPLIFIED:

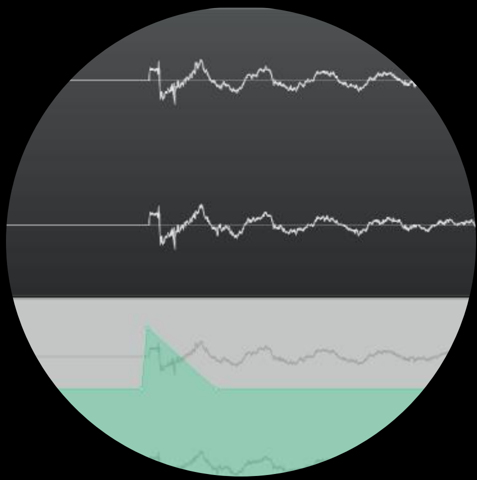
MANUAL MANIPULATION

WITH AUTOMATION: Using volume automation instead of clip gain for this purpose allows for *much* a greater level of control and customisation, allowing you to tweak the exact curve and duration of the boost/cut, and quickly compare different options to achieve the best result.

Again, there's no "right" or "wrong" here. Generally speaking, shorter, sharper boosts (like the middle one in the image above) will deliver a "snappier" sound, while experimenting with slightly longer, more "linear" curves will better preserve the original timbre, and can be elongated to also add additional low-end "punch" to a sound.

PS. Another benefit of doing this via automation, is that once you've found "the best" option for a given source, you can simply copy/paste it for every hit!

Super punchy!



TRANSIENTS. SIMPLIFIED: AUTOMATED SOLUTIONS

Beyond altering transients manually, you can, of course, also just use various plugins to get the job done "automatically" (albeit not as accurately):

WITH TRANSIENT DESIGNERS: Various plugins, like SPL's legendary "Transient Designer", allow you to boost or decrease the attack (or sustain) of an incoming source with a single knob. These tools offer a quick and easy solution for ADSR manipulation, with the only downside being they aren't always 100% accurate, and can miss certain hits and transients when fed complex material (which, is why I often prefer the manual route).

WITH EQ & COMPRESSION: As mentioned briefly in our snare drum example at the start of this chapter, you can always just use EQ (with a high-shelf boost), or Compression (with a slow attack setting) to boost transients in a pinch. Just keep in mind that EQ will boost the highs of the *entire* signal (not just of the transient), and Compression will unavoidably also boost sustain & noise.

TRANSIENTS. CONCLUSION.

Traditionally, EQ and Compression have always been the "go-to" options for engineers looking to "make something smack harder".

The thing is, you have to remember that "back in the day", that was *all they had* - There *were* no digital editing grids or transient designers which could process the "A" independantly from the "DSR"...

A NEW HOPE

While EQ and Compression are *still* great tools for the job, they aren't always *the best* tool for the job.

Try experimenting with manipulating your transients separately from the rest of the signal - Chances are, in a lot of cases, you'll end up with a MUCH cleaner result, which when applied across an entire mix, can make a HUGE difference!

GATING.
SIMPLIFIED.

"Gating" and "Expansion" are WITHOUT A DOUBT the most overlooked and underused tools in mixing.

When most engineers think gating, they just think "drum bleed removal"...

Crickets chirp

...When I think about gating, I think about dynamic frequency boosts, multiband ADSR enhancement, A TON OF "naughty" sidechaining & parallel tricks *(which would bring most engineers to tears)*, and then, *finally*, bleed removal...

There's just SO MUCH you can do with a gate/expander *beyond* just removing bleed and noise! It's almost CRIMINAL how little time in the limelight these AWESOME tools get...

Anyway... Enough talk! Let's take a look at what all this fuss is about...

GATING. SIMPLIFIED:

GATING VS. EXPANSION WHAT'S THE DIFFERENCE?

Just like it says on the tin, a "Gate" (in the context of audio), acts as a kind of "gate" or "bottleneck", which either *allows* audio to pass through (and be audible), or *prevents* audio from passing through (by muting it), depending on how high/low the "Threshold" control is set.

An "Expander" (or "Downward Expander", in this usage context), is essentially the "softer" alternative to a gate, with a lower ratio and "reduction range" control which allows you to *reduce* anything which falls below the pre-determined threshold rather than totally *muting* it. Typically, most gating plugins can also operate as expanders with a few small tweaks.

GATING. SIMPLIFIED:

THE BASICS

The controls on a typical gate are very similar to those of a compressor:

(PS. I'm using Fabfilter's PRO-G gate for these examples)

The "Threshold" control determines the *minimum volume level* at which the gate will "open" (unmute) and allow audio to pass through.

The "Ratio" (which, if used at *lower* ratios effectively turns the gate into an "Expander"), determines the *severity* of the reduction effect applied.

The "Range" control sets the minimum "volume floor" which any audio that falls *below* the threshold can be *reduced-to*.

GATING. SIMPLIFIED:

THE BASICS

Similarly to a compressor, we also have our standard "Attack" and "Release" knobs, but this time, with the addition of some new "Hold", "Knee" and "Lookahead" knobs.

On a gate/expander the Attack determines *how quickly* the gate will *open* once an incoming peak/signal has gone *above* the pre-determined threshold. Typically, fast attack times are preferable here to assure that the gate "opens" instantaneously, preventing it from "swallowing" any incoming transients.

Once said audio has gone back *below* the threshold, the Release control determines *how long* it'll take for the gate to return to its closed (muted/volume-reduced) state. When gating a snare, for example, you can use the *release* to adjust *how long* the sustain is allowed to "ring-out" for.

GATING. SIMPLIFIED:

THE BASICS

As for our 3 new controls:

The Hold control essentially acts as a buffer between the attack and release, letting us specify a "guaranteed" number of milliseconds that the gate is "forced" to stay open-for each time it's triggered, before ultimately cycling through to the release.

Lookahead, in layman's terms, tells the gate to "look ahead" in time, *foresee* incoming transients, and open *X ms too early* to avoid "missing" them. Super useful!

Finally, the Knee control can be thought of as a "severity" control for how the gating/expansion effect sounds. Generally speaking, "harder" knee settings sound more "rigid" or "robotic", while "softer" knee settings sound more "natural".

Phew! With all that out of the way, let's take a look at some usage scenarios...

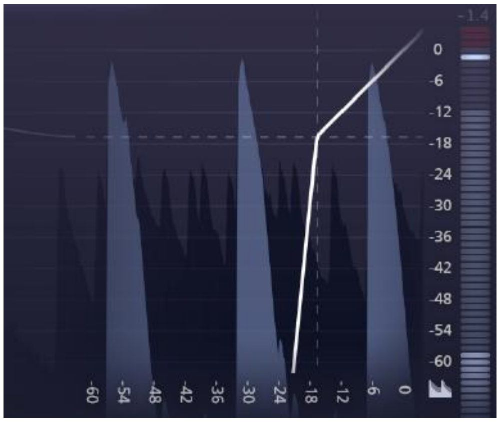
GATING. SIMPLIFIED:

USAGE SCENARIOS

BLEED/NOISE SUPPRESSION:

As mentioned previously, the most common usage for a gate/expander is to *eliminate* or *reduce* unwanted noise or bleed within a signal.

To do this with something like a live kick drum recording, start by setting the *attack*, *release*, and *hold* (if present) to their *fastest* settings, and the *threshold* control to its *highest* setting. At this point, the gate *should* be muting the signal entirely, as no peaks will be loud enough to trigger the gate to open.



Next, start *lowering* the threshold until *just* the kicks (which, if recorded well, *should* be significantly louder than the surrounding bleed), are triggering the gate to open and close momentarily for each hit. While setting the threshold as *high* as possible is preferable to prevent the bleed (snare and tom hits) from also opening the gate, make sure to set it low enough to account for dynamic changes in the performance, so that any quieter kick hits aren't being "missed".

GATING. SIMPLIFIED:

USAGE SCENARIOS

BLEED/NOISE SUPPRESSION CONTINUED:

Once you're sure that *only* the kick drum transients are triggering the gate to open, and that no lower-level kick hits are being overlooked, you can *slow down* the *hold* and/or *release* controls to keep the gate open for longer, ideally letting the full tail/sustain of each kick ring out more naturally, before the gate finally closes (mutes) again.

Congratulations! You've successfuly "gated" your first kick drum! At this point, you can experiment with slowing-down the *lookahead* control by a few milliseconds if you feel like the kicks are losing a bit of attack, or "softening" the *ratio* and/or *range* controls somewhat if you'd rather just *reduce* the bleed instead of totally *muting* it (effectively switching from *gating* to *expansion*, which can also help produce more "natural" sounding results).

Well. That's the most "typical" use for gating. Now onto the more EXCITING stuff...

GATING. SIMPLIFIED:

USAGE SCENARIOS

ADSR (ATTACK/DECAY/SUSTAIN/RELEASE) MANIPULATION:

Gates are great at removing the stuff that you *don't* want, but they're also *fantastic* for shaping the leftover stuff that you *do* want.

Let's say you have an electronic kick sample, for instance. It's sounding OK, but isn't really cutting through the other instruments, and is also ringing out for a bit too long, muddying up the mix as a result. Using a gate we can:

Enhance The Attack: Duplicate the kick channel and insert a gate plugin on it. Set the *attack*, *hold*, and *release* settings to *zero* (fastest), and bring down the *threshold* so that you're *just* hearing the sharp transient attack of each kick. Blend this channel in under the original in parallel to enhance the attack. Done!

Shorten The Sustain: Set up the gate just as we did in the "Bleed/Noise Suppression" section on the previous 2 pages. Then, use a combination of the *hold* and/or *release* controls to "tighten-up" and shorten the sustain of the kick to better match the song.

GATING. SIMPLIFIED:

USAGE SCENARIOS

"UPWARDS EXPANSION":

All of the tricks we've talked about so far have utilised gating/expansion in a "reductive" manner (as is probably the case with 99% of ALL gate usage throughout the history of mixing).

The thing is, for my money, the real bang for buck comes from using gating (or *expansion* in this case, to be more precise) in an "additive" configuration, also known as "Upwards Expansion".

When an expander is in an "upwards" configuration, any signal which goes *above* the pre-determined threshold will be *boosted* in volume based on the *ratio, range, attack* and *release* settings which have been dialled-in.

How is that useful? Well. Let's take a look...

GATING. SIMPLIFIED:

USAGE SCENARIOS

"UPWARDS EXPANSION" CONTINUED:

Remember how we talked about "Transient Designers" in the "Transients. Simplified." chapter of this book? Well, an upwards expander is essentially just a *really powerful, SUPER-customisable* transient designer.

For this purpose, my go-to plugin of choice is Fabfilter's PRO-MB Multi-Band Compressor (which also has gating & expansion functionality).

PS. Since we haven't covered the topic of "Multi-Band" processing in this book so far: "Multi-Band Processing" is when you divide a single, "full-frequency-range" sound into multiple "frequency bands" (bass, mids and treble, for example) which can be processed separately and independantly from each other.

GATING. SIMPLIFIED:

USAGE SCENARIOS

"UPWARDS EXPANSION" CONTINUED:

The combination of *upwards expansion* in a *multi-band* configuration, paired with other tools and techniques like *side-chaining, MIDI,* and *parallel processing* is equivalent to opening the "Pandora's Box" of mixing, and allows you to do SO MANY cool things that you *simply can't* through traditional means.

I can't cover *all* of these things in this book, as there's potentially an *endless* number of them, limited only by your creativity. However, I can give you a few good examples of how you can use Upwards Expansion in a mix:

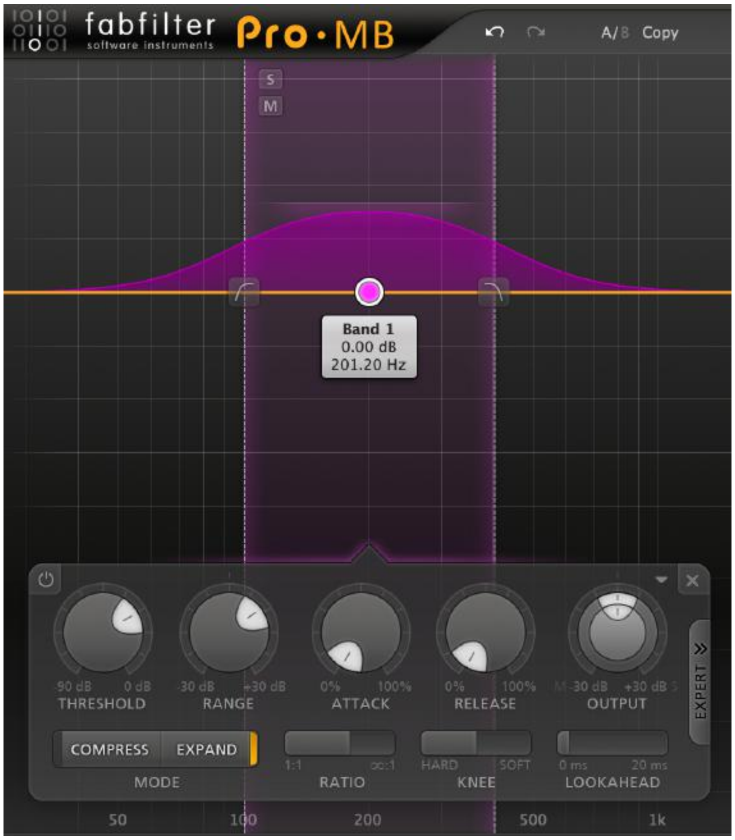
Like a "Dynamic EQ": Let's say you have a stereo drum loop where the kick, snare, hi-hats, and everything else are all baked-into a single stereo track. You may want to boost some low-mids into the snare, but doing so via EQ also affects the kick, making it sound "muddy" in the process...

GATING. SIMPLIFIED:

USAGE SCENARIOS

...If instead, you instanciate a band of *multi-Band Upwards Expansion* in that area using a plugin like PRO-MB and *side-chain* that band *internally* to the fundamental frequency of the snare, then you can adjust the *threshold, range, ratio, attack* and *release* controls so that the boost is *only* triggered by the snare hits, and *only* sticks around for as long as needed.

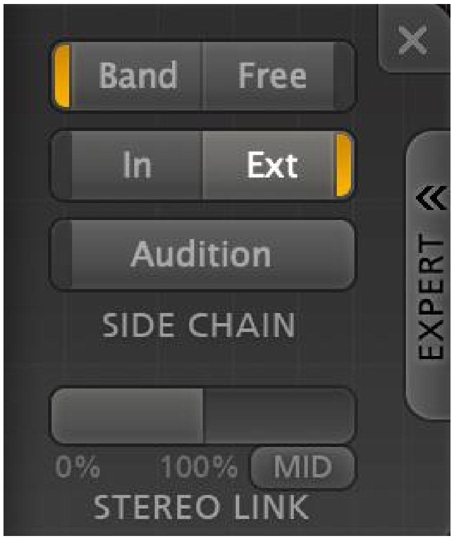
Voila! Dynamic, input dependant frequency boosting!



What's that?
You want *another* trick?
Oh, go on then...

GATING. SIMPLIFIED:

USAGE SCENARIOS



To Enhance Attack On Guitars : When dealing with heavily-distorted bass or electric guitars in a rock or metal context, you may want to enhance their front-end pick attack to help them better cut through the mix and really "punch" people in the eardrums.

The problem is, due to their distorted and "brickwalled" nature, these kinds of tracks typically have *little to no* transient information *to* work with, meaning most typical dynamic processing methods are totally off the table...

The solution? When recording heavy guitars, make sure to also always record a parallel duplicate of the *DI signal* alongside the "amped" signal. Since the raw, unprocessed performance will retain all of it's dynamic information, it can later be used as a "dummy" sidechain input for whatever dynamic processing you'd like to apply to the main, amped channel.

Somewhat confusing?
Here's a step-by-step example...

USAGE SCENARIOS

Enhancing Attack On Distorted Guitars With Upwards Expansion (Continued):

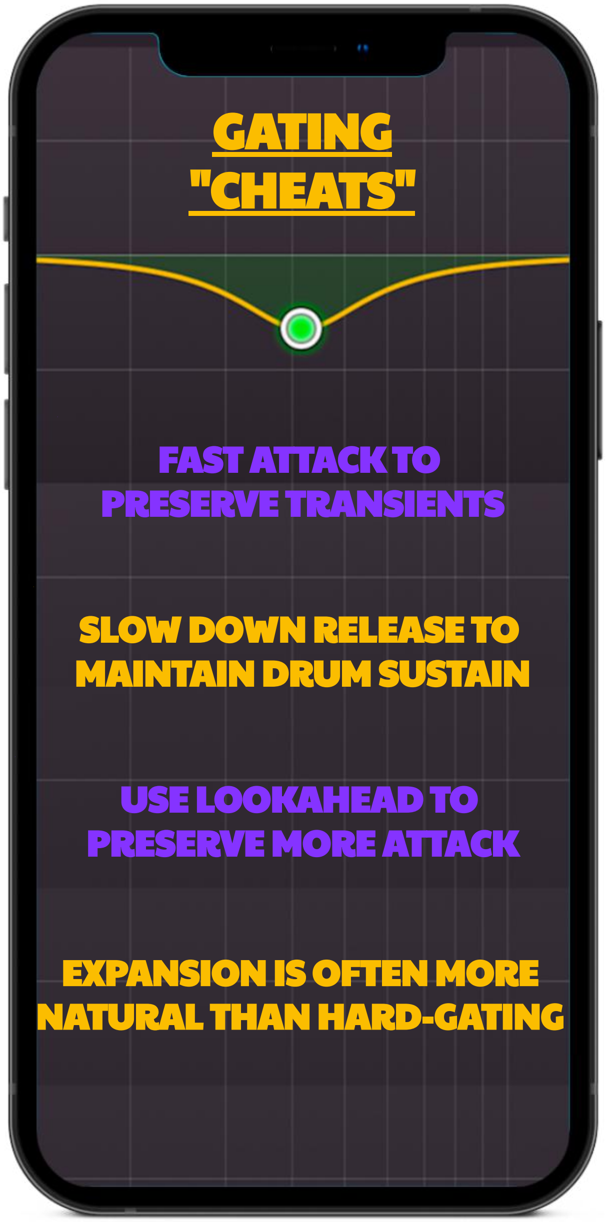
Step 1: Make sure you have your two channels ready (the amped guitar track and the DI guitar track), and that they are *identical* performances (any editing applied to one should have also been applied to the other simultaneously). If you're already working with a DI + Amp Sim for your "Amped" tone, you can simply *duplicate* the DI to another channel for this purpose.

Step 2: Send the DI channel to the "Amped" channel as a "sidechain" auxillary input, and make sure the DI *isn't* audible through the master buss/output.

Step 3: Insert your expander of choice on the amped guitar track, and set the input for the plugin/expansion band to external/sidechain/auxillary. At this point, the expander will now process the *amped track*, but do so according the the dynamics of *the raw/DI signal coming from the sidechain input*.

Step 4: Set the attack and release to their fastest settings and tweak the ratio and threshold until you're getting the desired "attack enhancement" effect.

GATING. CONCLUSION.



Hopefully everything we've talked about in this chapter will inspire you to give a little more love to gating and expansion in your future mixes.

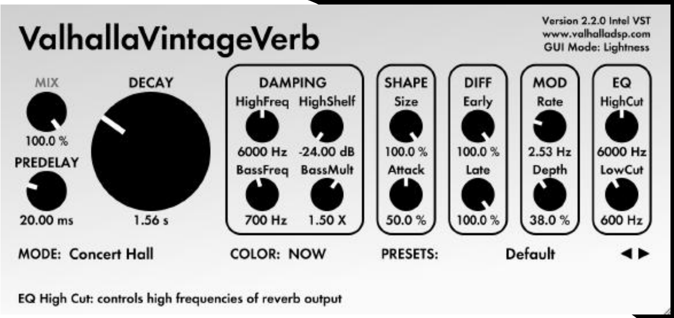
<u>There are just SO many unique and creative ways to use these tools to solve otherwise "unsolvable" audio issues. I feel like it would be such a huge shame if you were to miss out...</u>

OPEN THE GATES!

Sure. It can take a while to *really* get used to a new tool if you haven't used it much in the past, but if you just put in the practice, I GUARANTEE the benefits will be worth it!

<u>SO - Try experimenting with both downwards and upwards gating/expansion in your next project, and see what cool tricks and solutions to issues you can come up with. You never know, you may just come up with "the next big thing" in audio...</u>

REVERB.
SIMPLIFIED.

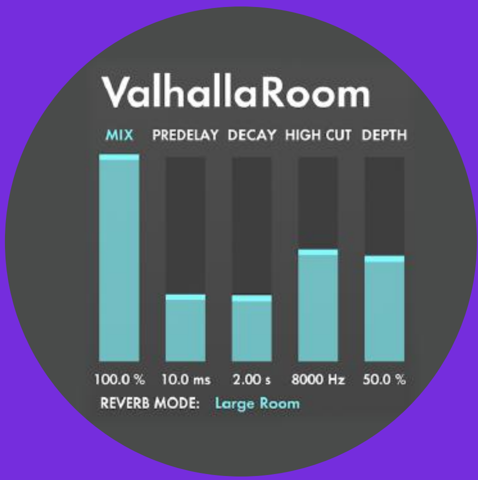


"Reverb" is one of the most iconic mixing effects out there. Everyone knows *what it is*, but not everyone knows *how to use it correctly...*

Just in case you're unfamiliar with the concept: "Reverb" is essentially a "time-based effect" which allows you to emulate the "Reverberation" properties that sounds produce when played within an acoustic space (a church, room, etc...).

By sending our "dry" recordings which were *ideally* tracked in an "acoustically dead", "neutral", "reflection-free" studio environment into reverbs, we can artificially place them within a *totally* different setting, achieving "larger than life", immersive results which *may* better-match the musical vibe or context we're working within.

So. Let's take a look at *how* we can set up a reverb effect to get the best results...

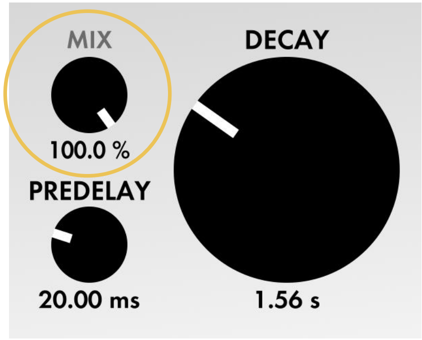


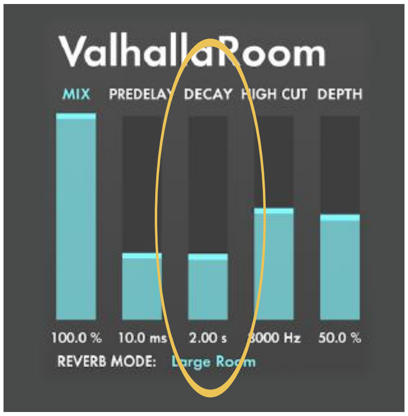
REVERB. SIMPLIFIED:

THE BASICS

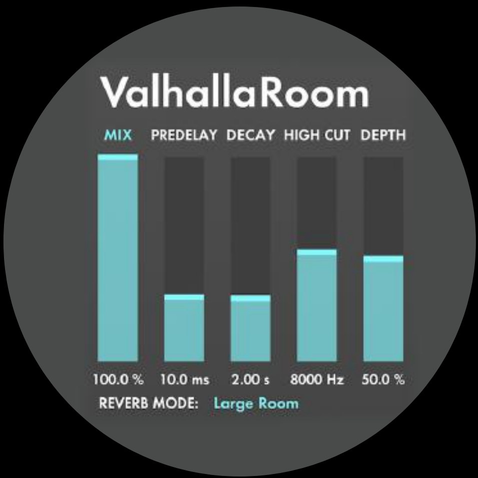
The selection of controls available on a reverb can change drastically from plugin to plugin (and hardware to hardware unit), but generally speaking, here are some of the primary knobs and faders you'll come across:

Mix/Dry-Wet: The "Mix" or "Dry/Wet" control adjusts the volume balance ratio between the "dry" (uneffected) signal being received by the reverb, and the "wet" (effected) signal being generated by the reverb.





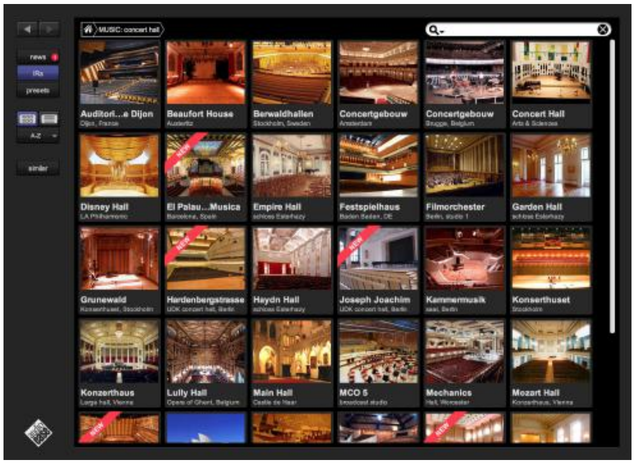
Decay/Length/Time/Tail: Although this setting can be called a plethora of different names, they all actually mean (and do) the exact same thing. Most commonly known as "Decay", this control simply adjusts the *length* of the reverb, or how long it will decay-for (in seconds) before fading out to silence. Smaller spaces tend to have shorter decay times, while larger spaces will have longer decay times.



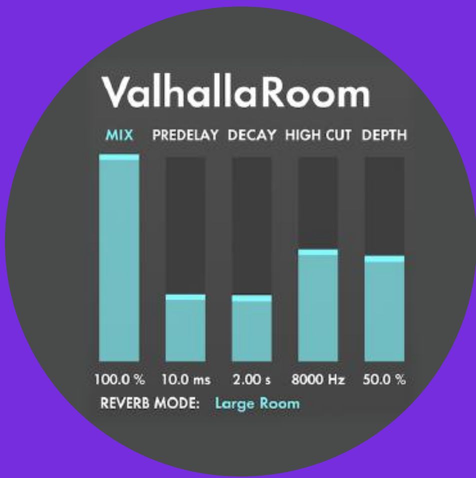
REVERB. SIMPLIFIED:

THE BASICS

Pre-Delay: The "Pre-Delay" knob allows you to "delay" the reverb by *X number of milliseconds*, adding a small time gap between the dry input signal and the wet effect return. This is incredibly useful, as the slight separation can prevent the reverb from "smearing" or "masking" the clarity and definition of the dry performance, resulting in a "cleaner" overall sound.

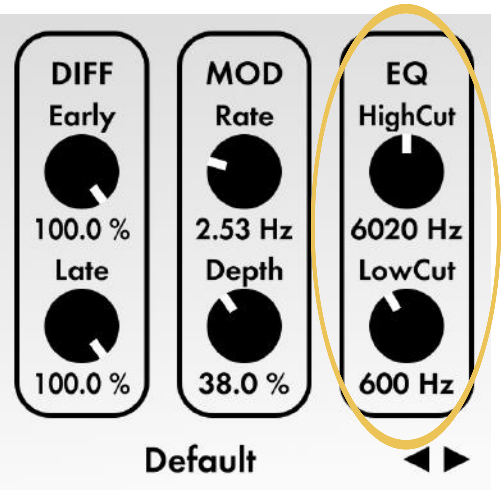


Mode/Type/Space: Most commonly referred-to as the "Mode" of a reverb, this setting allows you to select the general "type" of acoustic space or reverberation technology which the effect is emulating. Some of the most popular "reverb modes" include *halls, rooms, chambers, plates, springs* and *"linear" digital options*.



REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

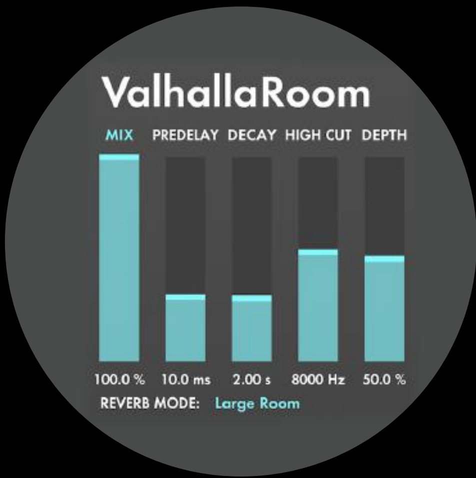


Low-High EQ/Damping/Filtering: Most reverbs will have a built in EQ and/or "Dampening" section which allows you to shape the overall tone of the effect. Generally speaking, reducing some of the highs and lows of a reverb can allow it to sit more comfortably within a mix.

Beyond these "standard" features, a few of the other controls you may find on different reverb units are "Size", "Diffusion" and "Modulation", which all allow you to further customise the tone and vibe of the effect.

It's important to note that there's *no such thing* as a "correct" reverb setting, only what *works best* for a given musical context.

That being said, there *are* a few "best practices" you can use to make the most of your reverbs, which we can take a look at now...

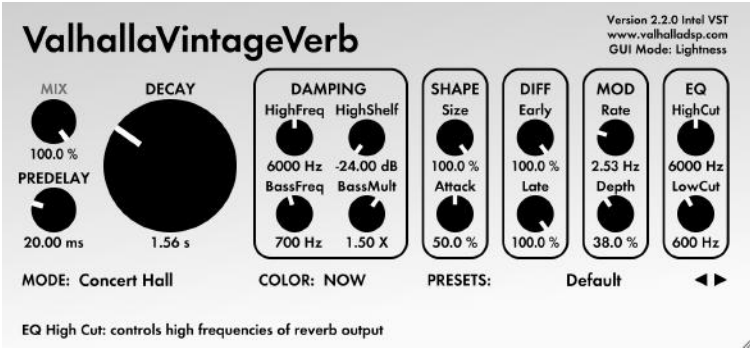


REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

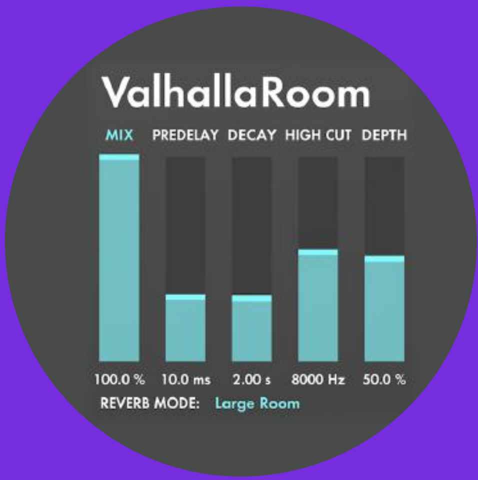
[Genre, tempo and many other context-specific factors come into play when determining the "best" reverb approach to go with for a particular instrument or song.]

This doesn't mean that you should go and splash out on a dozen different reverb units, but it *does* mean that it's a good idea to invest in a really *flexible* "jack of all trades" reverb early on if you're looking to achieve professional results from the get-go.



Valhalla DSP's "VintageVerb", for example, is an extremely versatile, tweakable, affordable industry-standard reverb which has been used on *thousands* of hit records. I've been using it as my primary reverb for over a decade at this point, and couldn't be happier with the results!

Lets take a look at EXACTLY how I set it up in my mixes...



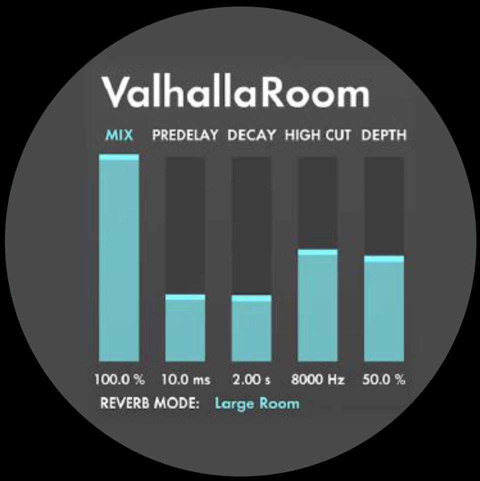
REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

Choosing The "Right" Reverb Mode/Space:

Unless you're *specifically* looking to make a song sound like it was recorded within a certain "realistic" acoustic space (like placing a live, solo acoustic singer/songwriter performance within a nice cathedral or studio space, for example), for 90% of your everyday mixing needs, you simply *can't* go wrong with a nice "Concert Hall" emulation.

The thing is, "Realism" *doesn't* always equate to "the most exciting" or "most suitable". If you were to place every single element of a song into different "Realistic" acoustic spaces, for example, you'd most likely just end up with a very "disconnected" sounding mix, in which nothing is "gelling together" or "cohesive"...



REVERB. SIMPLIFIED:

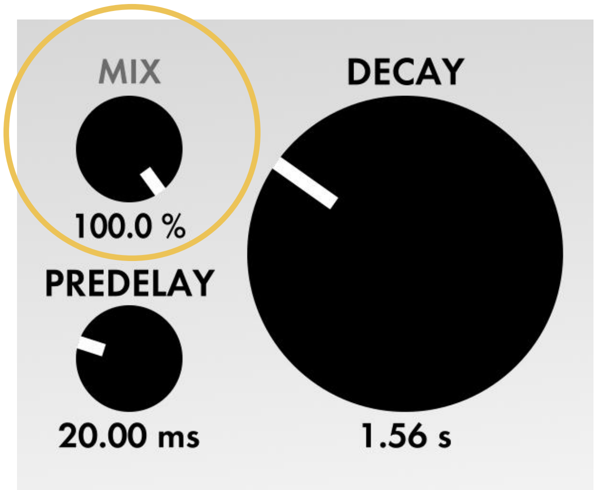
REVERB TIPS & TRICKS

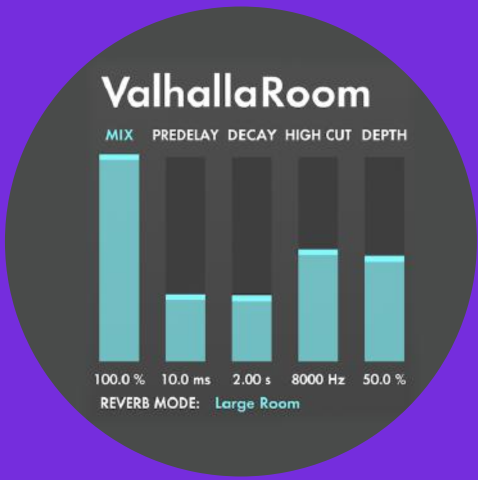
Using the Wet-Dry/Mix knob and setting up your Reverb as a "Send":

As mentioned previously, when placing a reverb plugin directly on the instrument channel you'd like to effect, you can use the "Wet/Dry" or "Mix" knob to adjust the ratio of the effect to that of the dry signal. The problem is, using reverb in this "Direct Insert" configuration means you *can't* process the effect separately if required, and every change you make to the reverb amount will also affect the volume of that instrument within the mix.

For these reasons, effects which have dry/wet functionality (like reverbs and delays) are often better used as "Sends" on totally separate, independant channels.

To create a "Reverb Send" (also known as a "Reverb Buss", in this context), create a new audio channel within your DAW, insert your reverb plugin of choice, and set the wet-dry/mix knob to 100% wet...





REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

...Now that your reverb send/buss has been created, you can send individual instrument channels into said channel at *different* "send levels" (amounts) using the "Routing" or "I/O" (input and output) functionality of the channels in your DAW. The exact process for achieving this and terminology used may vary slightly depending on the software/program you're using, but rest assured that you *can* do it within any one of them.

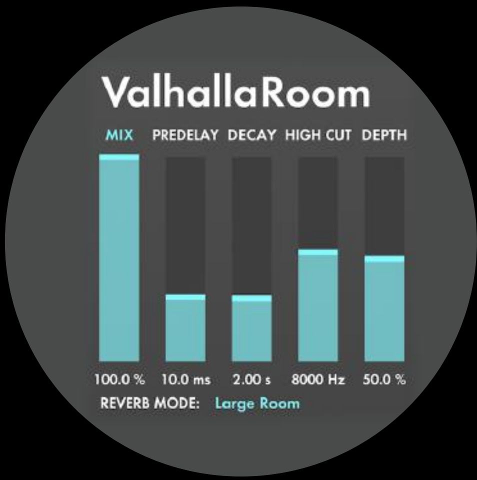
Some of the many benefits of using "Sends" over "Direct Inserts" for these types of effects include:

- Maintaining instrument volume balances when changing "reverb send levels"

- Saving CPU by using the same, single reverb instance for multiple instruments

- The ability to process the "reverb buss" as a standalone channel with EQ, etc.

- Using the same reverb for all instruments often sounds more cohesive



REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

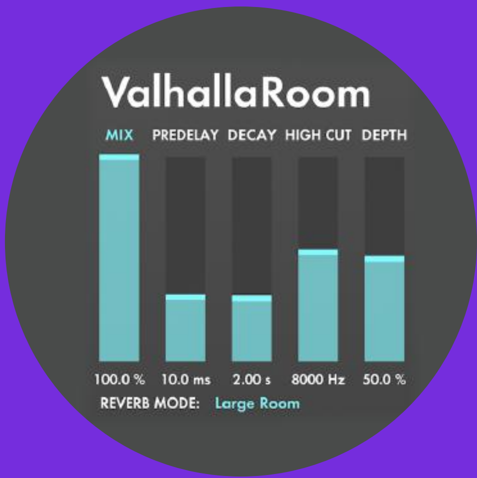
Now that the "setup" stage is complete, and the reverb is ready to be used, let's take a look at how you can tweak the various controls to achieve the most suitable results.

Adjusting your reverb settings to sound "right" for your mix:

Setting The Decay: First off, you'll want to adjust the length or "Decay" of the reverb to fit the particular BPM/Tempo of the song you're working on. Luckily, there's a basic mathematical formula for this which will provide you with a few "logical" options to choose from:

60000 (milliseconds/ms per-minute) ÷ BPM (of the song) = X ms per ¼-note beat

Once you've calculated the number of milliseconds per quarter note beat for the particular song you're working on, you can simply multiply this number in increments of 2 to reach half notes, 1 bar, etc...



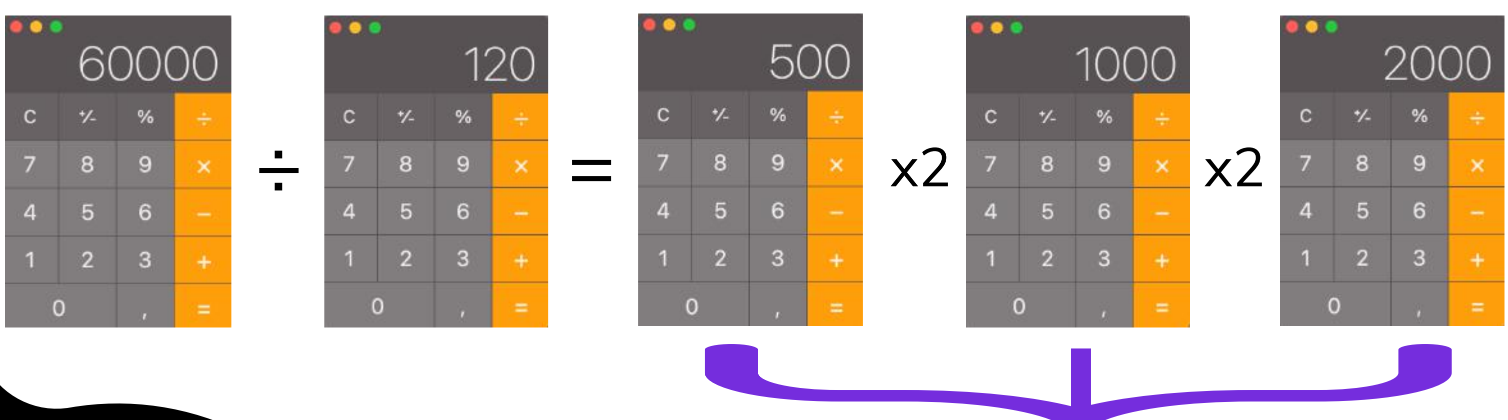
REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

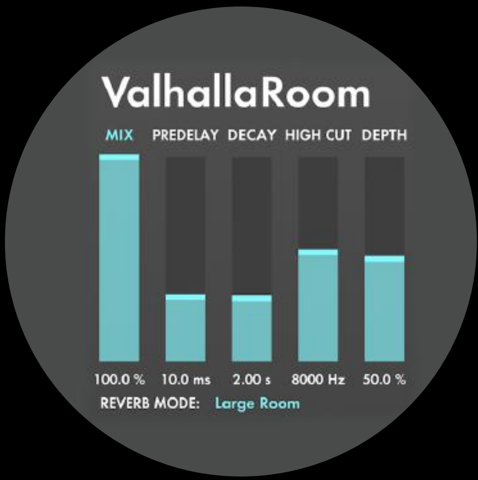
...While there's no such thing as a "correct" decay time, generally speaking, a reverb length of 4 beats (1 bar), or somewhere between 1-3 seconds *tends* to work well in most scenarios.

Another way of thinking of it, is that you *ideally* want something like a snare reverb to finish reverberating *just* as the next snare hit is coming around, as to not "stack" them on top of each other and contribute to unnecessary build-ups.

Basically: Just use the formula from the previous page, set the resulting number as your reverb's decay, listen, and multiply that number by 2 until you find the length that works for your particular context. Here's an example at 120BPM:



(Pick one of these 3)



REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

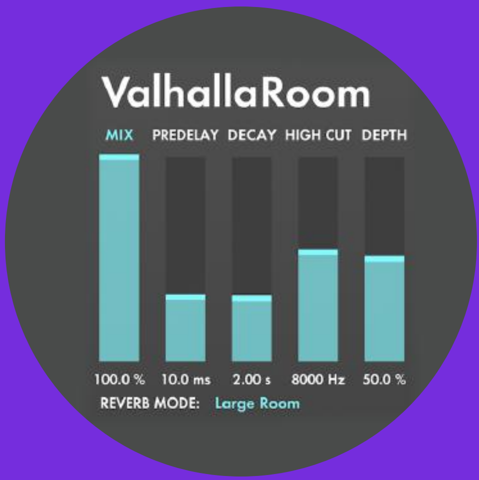
Awesome! So far, we've sorted the *type* and *length* of our reverb. It's sounding *pretty* good, and is (hopefully) only as long enough as it *needs to be* as to not make our mix sound "muddy" or "chaotic".

A lot of amateur mixers who don't know any better would probably call it a day at this point and move onto something else, but believe it or not, this reverb is still a *long* way from sounding "professional" - Guess we better keep going then!

Setting the Pre-Delay: As mentioned previously, the "Pre-Delay" knob on a reverb allows us to "separate" the "dry" input signal and "wet" reverberated signal from each other by *X number of milliseconds* in order to differentiate the two sounds slightly and prevent "masking".

Again, while there's no "correct" Pre-Delay setting, logically (and musically), it makes sense to *also* time this setting to a "rhythmic subdivision" of the BPM/Tempo.

(Albeit, a very *short* subdivision...)



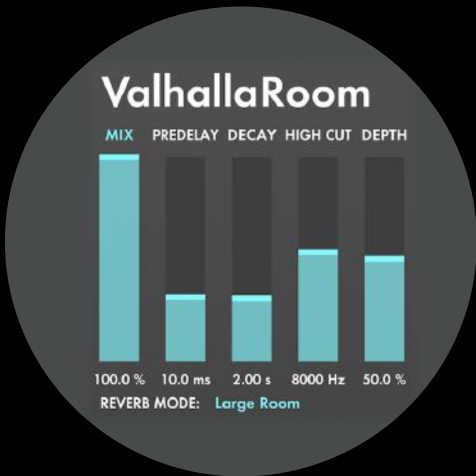
REVERB. SIMPLIFIED:

REVERB TIPS & TRICKS

...Luckily, to do this, we're mostly just going to be be using the same formula as before, except that this time, instead of *multiplying* our starting quarter-note value by 2 to reach *multiple note* values, we'll be *dividing* it by 2 to reach *shorter/faster* note subdivisions.

Generally speaking, Pre-Delay times of 10-30ms will provide enough dry/wet signal separation to do the trick while not making the reverb sound blatantly "delayed". Here's an example of the formula in action:

This being said, world-renowned mixer Chris Lord-Alge (Green Day, Paramore) famously uses a whopping 150ms of pre-delay for his go-to Lexicon 480 Vocal Plate setting. So don't be afraid to experiment with longer times too!



REVERB. SIMPLIFIED:
REVERB TIPS & TRICKS

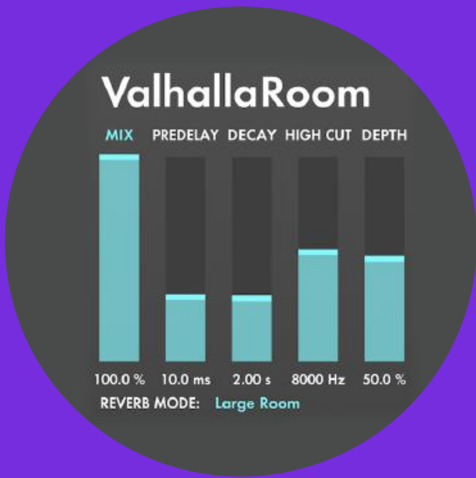
At this point, we've set up all of our "reverb specific" settings. But we're not done *just* yet! There's one more *incredibly* powerful trick we can use to make our reverbs sound *even better.*

Mixing is all about the "finite" balancing of volume & frequency, and the masterful "juggling" of space. In a "well constructed" mix, every sound and instrument will "live" comfortably in it's own "purpose specific" range, resulting in maximum clarity, audibility, impact and punch.

If, into this "finite balance", you introduce a bunch of reverb effects which have long, sustained, "boomy" low-end and annoying, harsh, "fizzy" high-end, you're just undoing a lot of that hard work...

But fret not! This is *exactly* why God invented equalisers!

Let's take a look at how you can use them to "slimline" your Reverbs...



REVERB. SIMPLIFIED:

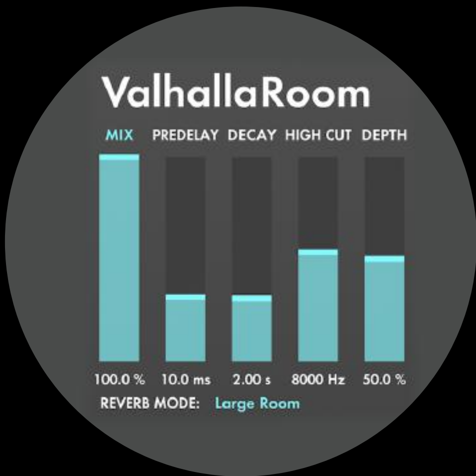
REVERB TIPS & TRICKS

EQ'ing/Filtering your Reverbs: Based on everything we've just talked about, it's often a good idea to filter out *quite a lot* of the low-lows and high-highs from your reverbs.

This "diet plan" essentially serves the purpose of "channeling" reverb further into the mid-range, which, funnily enough, both makes it *more audible* in a mix AND *less "bloated" at the same time!*

A good starting point when filtering reverbs is the classic "Abbey Road Reverb Trick".

Basically, "back in the day", the engineers working at Abbey Road Studios (most famous for being the place where The Beatles recorded their albums), figured out that by filtering out the lows and highs of their reverbs at 600Hz and 8kHz respectively, they could achieve a *much* more "natural" sound that works *much* better within a mix.



REVERB TIPS & TRICKS

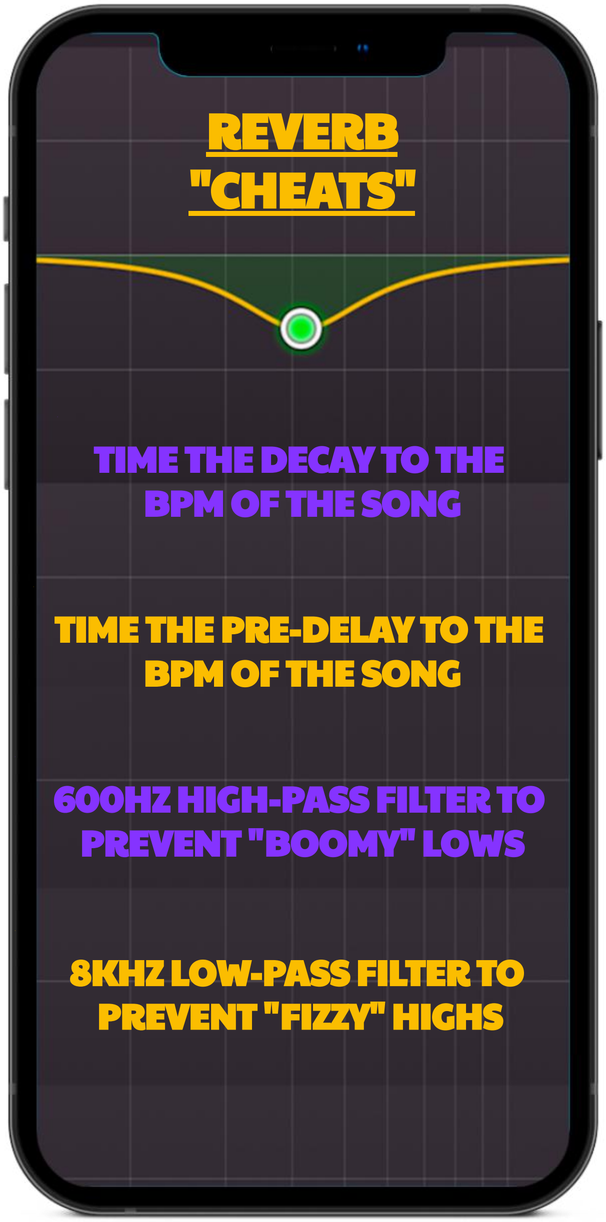
While the *exact* frequencies stated in the "Abbey Road Trick" *will* work perfectly well in 90% of cases, and you can safely just copy them straight into your own mixes as-is with zero tweaks, doubts or worries, there *are* a few things you should still keep in mind:

Mix "Density": A "dense" mix with *dozens* of instruments and sounds competing for space will *definitely* benefit from "slimlined" reverb sounds, but a "sparse" mix with only *a few* elements may actually *benefit* from a "bolder" reverb to fill-up some of the otherwise empty space.

All Instruments/Sounds Are Different: 600Hz and 8kHz are low and high enough frequencies to resolve *most* of the standard "boominess" and "fizziness" problems you'll come across. However... *Every sound/instrument is different*, so don't be afraid to *lower* or *increase* these "starting points" if you feel like a particular case requires more or less of either.

"Pre" or "Post" Filtering?: For the most "natural" results, you'll actually want to place your EQ/Filters *before* your reverb plugin rather than *after.* This way, you're essentially *preventing* the problem of the lows and highs reverberating instead of "creating a problem and fixing it". (It just makes more sense this way around...)

REVERB. CONCLUSION.



The difference in sound quality between a random "raw" reverb and one which has been "properly dialled-in" using all of the tips and tricks we've discussed in this chapter is *staggering* - and being that it's *such* an important effect in the grand scheme of mixing, it's one of those things which you just *really need to get right* to be taken seriously in the field.

TREAT REVERB LIKE ANOTHER INSTRUMENT!

Essentially, you just want to think of your effect sends as additional instruments which have been added into the song.

They *aren't* exempt from EQ and other forms of processing, so just feel free to do whatever it takes to help them reach their full sonic potential!

DELAY.

SIMPLIFIED.

"Delay", "Delay", "Delay", is, is, is, the, the, the, effect, effect, effect, which, which, which, creates, creates, creates, "echoes", "echoes", "echoes"...

Reverb is cool and all, but personally, I'm a HUGE sucker for "Delay". It's just *such a musical effect* in how it can transform even the most "plain" and boring sound or performance into something *much* more interesting and "rhythmically rich".

If you're not familiar with it: "Delay" or "Echo", is an audio effect which we can use to *multiply, delay* and *repeat* an incoming sound/instrument signal at user-specified time intervals.

To hear a blatant example of delay being used to great effect in popular music, just check out any song by the band U2 ("Pride" and "Where The Streets Have No Name" are a perfect place to start, as both songs heavily utilise the effect on electric guitars during their introductions).

Sounds good, right? Well, let's see how you can achieve these kinds of Delays in YOUR mixes...

DELAY. SIMPLIFIED: THE BASICS

Delay can be used in two primary ways: Firstly, as a "constant effect", to add a sense of "space and depth" or "rhythmic sophistication" to a part or instrument. Or secondly, as more of an "event", to *emphasize* important words, notes, or chords at *key moments* in a track.

Either way, setting it up "correctly" will go a long way towards making sure it fits into its role comfortably. On that note, lets take a look at *how* we can go about doing *just that*:

Beyond the "Mix/Wet-Dry" and "EQ/Filtering" knobs which we've already talked about previously in the "Reverb. Simplified." chapter, the three other primary settings you'll find on a delay are "Delay Time", "Feedback" and "Mode".

PS. Before we get into what these settings do, I'll also quickly note that as with Reverb, you'll primarily want to be using your Delay effects on their own "Delay Send" or "Delay Buss" channels at 100% wet rather than a "direct insert". Feel free to skip back to the previous chapter for a brief refresher on how to set all of that up if need be!

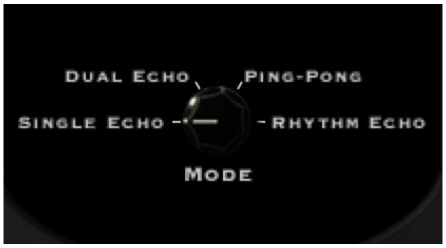
Anyway. Returning to our 3 primary delay controls...

DELAY. SIMPLIFIED: THE BASICS

"Delay Time": This control allows us to determine the *rhythmic interval* at which the incoming sound will be "delayed & reproduced" (16th notes, 8th notes, 4th notes, etc). Most modern delay plugins are *automatically synchronized* to the BPM of your DAW, but you can usually also adjust them *manually* if need be.

"Feedback": Also commonly referred to as "Repeats", the "Feedback" setting on a delay adjusts *how many times* the delayed signal will repeat and "feed back" into itself before stopping. Depending on the "sonic character" of the particular delay effect you're using, these repeats may either be reproduced *exactly* every time, *or,* with a slight "degradation" effect which intentionally "worsens" for each consecutive echo.

"Delay Mode": Delay comes in 3 primary "Modes": "Single", which is a mono delay that repeats identically in both of the L/R channels. "Dual", which allows you to choose different, independent delay intervals for left and right. And "Ping- Pong", in which a mono delay takes turns bouncing between L/R, becoming stereo as a result.

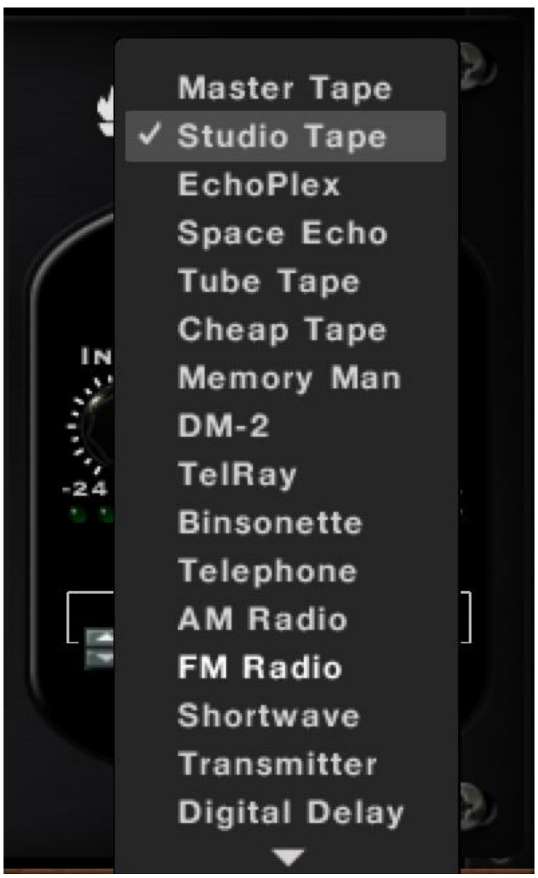


As usual, there's no right or wrong choice with any of these settings, only what *works best* for the particular effect you're trying to achieve.

DELAY. SIMPLIFIED:

THE BASICS

Aside from those 3 *primary* settings, depending on the particular delay you're using, you may also be given the option to choose the "style" or "character" of the effect.



The *fantastic* "EchoBoy" plugin by Soundtoys, for example, gives you access to *dozens* of famous delay sounds and hardware emulations which cover everything from "clean" digital delays to "compressed" tape delays to "lo-fi" telephone delays...

There's a reason I and a million other engineers have been using this plugin as our *go-to*, all-in-one delay solution for such a long time: It's just hands-down *the best there is!*

PS. While any one of these options can work well depending on context, for most "everyday mixing" purposes, you simply can't beat a good "Studio Tape" emulation for its "Compressed and Saturated" sound...

DELAY. SIMPLIFIED:

DELAY TIPS & TRICKS

Now that we've covered how all of the controls on a delay *work*, let's take a look at how we can go about *setting them up* for the "best" results.

As mentioned previously, there are a few different ways you can utilise delays within a mix. Let's take a look at a few of them and discuss some of the common practices used to get them sounding "right":

"Constant" Delays: Much like you might apply a "constant" reverb to a lead vocal for the full duration of a song, a "Constant Delay" (as I call it), is a delay which is *always present* for purpose of adding "space" and "width" to a sound.

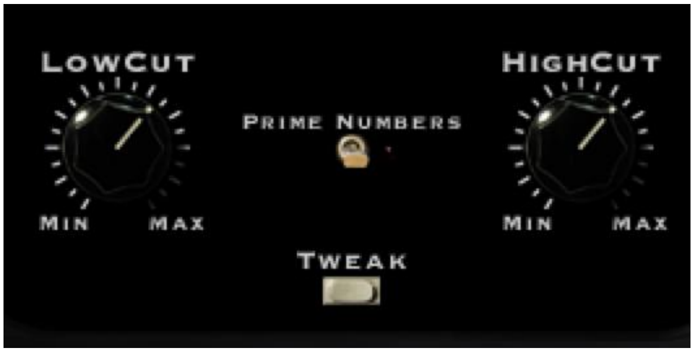
You can experiment with different delay "modes" and "characters" for this purpose, but personally, I *love* a stereo 4th/8th/16th-note "Ping-Pong" tape delay with a short feedback, and some fairly heavy EQ/filtering.

This effect is meant to be *subtly "felt"* rather than *blatantly "heard",* and just really helps add some nice "movement" and "depth" to something like a lead vocal.

Regarding the "feedback" and "filtering" settings...

DELAY. SIMPLIFIED:

DELAY TIPS & TRICKS



When it comes to EQ'ing/filtering delays, the exact same "Abbey Road" rules we talked about in "Reverb. Simplified" apply: Generally speaking, simply getting rid of everything below 600Hz and above 6-8kHz will do *wonders* towards making your delays take up *less* space, while also being *more audible* in a mix.

Similarly, a lot of what we talked about regarding adjusting reverb decay times also applies to adjusting delay feedback amounts: Essentially, you just want to make sure your delays aren't repeating for longer than they "need" to, and aren't contributing "mud" and "chaos" to the mix. Sometimes a single "delay repeat" is enough, while other times you may want to "complete" a bar for the sake of musicality (1 Original sound + 3 repeats = 4 total sound repetitions).

That covers "constant delays". Finally, let's take a look at "event" delays...

DELAY. SIMPLIFIED:

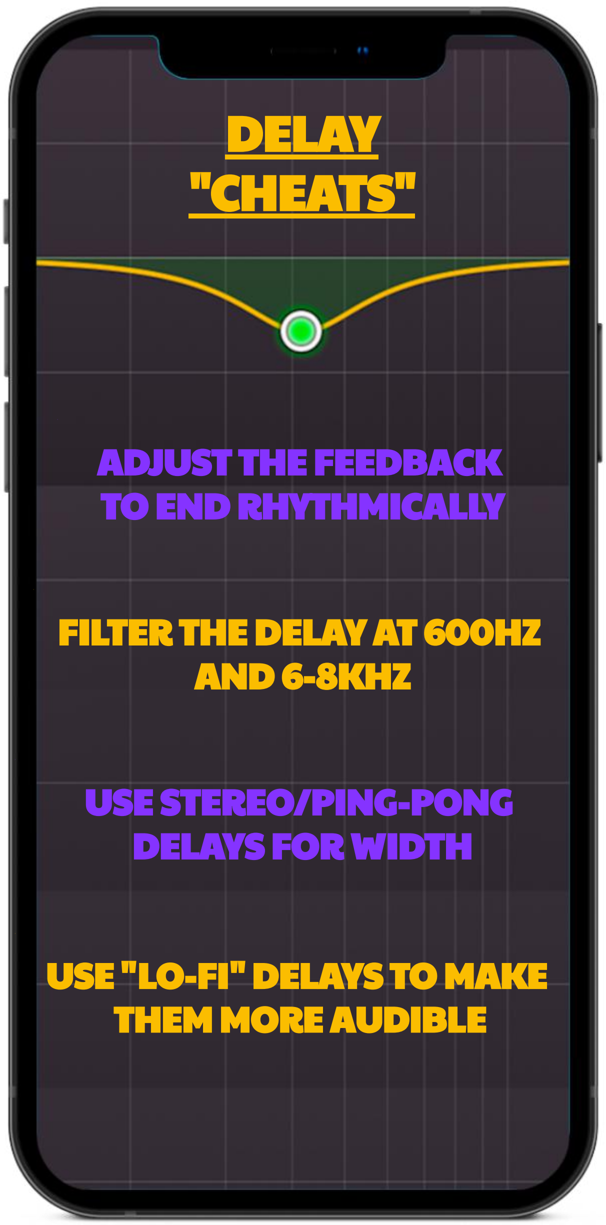
DELAY TIPS & TRICKS

"Event" Delays: If you listen to pretty much *any* professionally mixed song on the radio or the internet, I can pretty much *guarantee* you'll hear what I like to call "Event Delays" being used to strategically "emphasize" certain words in the lead vocal performance, or simply "fill up" empty gaps/spaces in a "musically pleasing" way.

In my own mixing, I'll typically have 4-5 "delay busses" (each with a different "time", "mode" and "character" setting) ready-to-go for the purpose of "sending" a particular word into any one of them via "Automation".

Just in case you've never heard of it: "Automation" is a DAW functionality which allows the user to "automate" certain channel or plugin parameters. In the context of delay, we can use automation to *auto-adjust* the levels at which an instrument is being sent to each delay buss at different points throughout the song (for example, the send to "delay buss 1" can be set to minus infinity for most of the song, but jump to -10dB momentarily to catch certain words which we'd like repeated).

DELAY. CONCLUSION.



Dialling-in various delays and then automating them to "go off" for certain words and sounds in a song is *literally* my *favourite* part of the *entire* mixing process.

It's just *so* satisfying hearing how then can *totally* transform and bring a track to life!

DELAY SEPARATES THE PROS FROM THE "SCHMOS"!

There's just *so* much room for creativity with this effect...

So... Don't sleep on it! Start experimenting with using delay in your own mixes, and you'll be amazed by how much of a difference it can make when used well!

FINAL WORDS

Hopefully mixing is a little less complicated for you now...

This journey is a long one. In fact, I'm not sure it *ever* really "ends"...

I *still* find myself learning new things and coming to new conclusions nearly *every single day*, even this far into my career!

ANYWAY...

The key, I've found, is to *never stop learning*, and *never* assume you finally "know it all", as *that's* the point where you stop growing and things start going downhill.

This, has been Mixing Simplified.

Please feel free to get in touch with ANY questions or queries you may have *whatsoever.*

Until next time...

Thomas Brett - Founder

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