MASTERING

WITH OZONE

2013 EDITION

PRINCIPLES, TIPS, *and* TECHNIQUES

iZotope

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INTENDED AUDIENCE FOR THIS GUIDE

If you don't know anything about mastering or mastering software, this Guide is a great place to start. Sure, iZotope thinks you should use iZotope **Ozone**™ (**www.izotope.com/ozone**) to master your audio…you certainly could. But iZotope has learned so much from the audio community over the past 10+ years that we're happy to give something back in return: a Guide that's useful for *everyone* who wants to learn more about mastering. As a result, this Guide can be freely copied or distributed for non-commercial purposes.

If you don't fully understand mastering but already have Ozone, this Guide can help you better understand the powerful sound-shaping tools at your disposal. Each chapter demonstrates many useful concepts that you can apply to your next mastering project. You can also follow along with the free 10-day trial of Ozone at **www.izotope.com/ozone**.

If you already have Ozone and already know the basics of mastering, this guide can show you new tricks or techniques that are possible in Ozone's modules. Just read through and say "Yeah, I knew that" when appropriate for the other parts.

ABOUT THE 2013 EDITION

The 2013 Edition of this Guide has been revised by **Jonathan Wyner**, Chief Mastering Engineer and founder of **M Works Mastering Studios** (**www.m-works.com**) in Cambridge, MA. Over the past 25 years, Jonathan has mastered more than 5000 CDs across every musical idiom (and some non-musical idioms as well). Notable mastering credits include Aerosmith, David Bowie, the Cream, Kiri Te Kanawa, Rahsaan Roland Kirk, Nirvana, the London Symphony Orchestra, Aimee Mann, Pink Floyd, Pete Seeger, Bruce Springsteen, Tiny Tim, and many more. In addition to his mastering projects, Jonathan regularly opens eyes and ears to the art of mastering as a faculty member at Berklee College of Music. iZotope is delighted to have Jonathan bring his respected perspectives and practical experience to the 2013 edition of this Guide.

ADDITIONAL RESOURCES

If you are interested in exploring mastering more in-depth, there are plenty of resources to explore, including *Audio Mastering: Essential Practices* (by Jonathan Wyner himself!) , which is available as of May 2013 from Berklee Press, both in hard copy and as an e-book for various platforms. For even more hands-on training from the comfort of your home, **Berkleemusic.com** offers an online course in mastering. Ultimately practice is where it's at, but having a firm platform of knowledge on which to stand when you're practicing your art is invaluable.

ABOUT iZOTOPE

iZotope is a research-driven audio technology company based in Cambridge, Massachusetts. Its award-winning products and audio technologies are used by millions of people in over 50 countries, from consumers to musicians to major film, TV, and radio studios. Learn more at **www.izotope.com**.

1: INTRODUCTION

Mastering is often thought of as a mysterious art form. This guide aims to tackle that mystery head on—to not just explain what mastering is, but to outline how one might go about achieving the primary goal of any good mastering engineer. And what's that primary goal? It's simple: to prepare an audio recording for distribution while ensuring it sounds at least as good (if not better!) when it goes out than it did when it came in. So let's get started!

IT ALL BEGINS WITH THE MIX

You've just finished mixing what you think is a pretty good recording. The playing is good, the recording is clean, and the mix is decent. You export a file or burn it to a CD and proudly pop it in your audio player. Yet when you hear it played next to a commercial track from your favorite artist, you think that somehow your recording is missing that sonic "X factor." You can't help but wonder… what's wrong with my recording?

DIAGNOSING COMMON PROBLEMS

- It sounds small, and isn't 'loud' enough. Turning it up or mixing down at a higher level doesn't solve the problem. Yes, that makes it sound louder, but doesn't add the required impact or clarity.
- It sounds dull. Other recordings are warm, deep yet bright and open, all at the same time. You try boosting the EQ at high frequencies, but now your song just sounds harsh and noisy.
- The instruments and vocals sound thin and lack the same sense of fullness that your favorite recordings have. You patch in a compressor and adjust some controls…and now the whole mix sounds squashed. The vocal might sound louder, but the cymbals have no dynamics. It's fuller… and lifeless.
- The bass doesn't have punch. You boost it with some low end EQ, but now it just sounds louder and muddier. Not punchier.
- You can hear all the instruments in your mix, and they all seem to have their own "place" in the stereo image, but the overall image sounds wrong. Other tracks have width and image that you just can't seem to get from panning the individual tracks.
- You had reverb on the individual tracks, but it just sounds like a bunch of instruments in a bunch of different spaces. Your other CDs have a sort of cohesive space that brings all the parts together. Not like rooms within a room, but a "spaciousness" that works across the entire mix.

WHAT DO I DO NOW?

Mastering is a process that can, and with practice often does, take recordings to the next level. It might be the process that can address the problems listed above. What mastering SHOULDN'T be expected to do is completely reinvent the sound of your recording. Mastering is not a substitute for good mixing, or good arranging for that matter! "Loud" records are a result of good writing/arranging/mixing AND mastering. They are made to sound GOOD and loud (if LOUD is what you are after) from the get-go, not just at the end. Once you have reached the final step of mixing with something that represents your best effort, something that you are proud of, then it's time to dig in and see how much further mastering can get you toward the sound that you hear in your mind's ear.

You won't become Bob Ludwig (see **www.gatewaymastering.com**) overnight, but you can make dramatic improvements to the quality of your mastered recordings with a little work.

In the end there are no right answers, no wrong answers, and no hard and fast rules. However, there are some well-known principles of audio production and mastering that are worth thinking through as you experiment. (And for goodness sake, have fun!)

2: WHAT IS MASTERING?

Although there are many definitions of what "mastering" is, for the purpose of this guide we refer to "mastering" as the process of taking a mix and preparing it for distribution. In general, this involves the following steps and goals.

THE SOUND OF A "RECORD"

The goal of this step is to take a good mix (usually in the form of a stereo file) and put the final touches on it. This can involve adjusting levels and general "sweetening" of the mix. Think of it as the difference between a good-sounding mix and a professional-sounding master. This process can, when necessary, involve adding things such as broad equalization, compression, limiting, etc. This process is often actually referred to as "premastering" in the world of LP and CD replication, but we're going to refer to it as mastering for simplicity.

CONSISTENCY ACROSS AN ALBUM

Consideration also has to be made for how the individual tracks of an album work together when played one after another. Is there a consistent sound? Are the levels matched? Does the collection have a common "character," or at least play back evenly so that the listener doesn't have to adjust the volume?

This process is generally included in the previous step, with the additional evaluation of how individual tracks sound in sequence and in relation to each other. This doesn't mean that you simply make one preset and use it on all your tracks so that they have a consistent sound. Instead, the goal is to reconcile the differences between tracks while maintaining (or even enhancing) the character of each of them, which will most likely mean different settings for different tracks.

PREPARATION FOR DISTRIBUTION

The final step usually involves preparing the song or sequence of songs for download, manufacturing and/or duplication/replication. This step varies depending on the intended delivery format. In the case of a CD, it can mean converting to 16 bit/44.1 kHz audio through resampling and/or dithering, and setting track indexes, track gaps, PQ codes, and other CD-specific markings. For web-centered distribution, you might need to adjust the levels to prepare for conversion to AAC, MP3 or hi-resolution files and include the required metadata.

OZONE TIP

Ozone is not designed to address these distribution-based tasks by itself, but is instead meant to work within dedicated Digital Audio Workstation (DAW) software to address the audio-specific portion of the mastering tasks. Supported DAWs include Avid Pro Tools; Steinberg Cubase, Nuendo, and WaveLab; Apple GarageBand and Logic; Adobe Audition and Premiere; Cakewalk SONAR; Sony ACID, Sound Forge, and Vegas; Cockos REAPER; Ableton Live, and many more.

APPROACHES TO MASTERING

There are three ways to come at mastering. Let's give these methods personalities and call them Ms. Fix-It, Mr. Make-It-Better, and Ms. Nuts-and-Bolts.

Ms. Fix-It

Ms. Fix-it is someone who can recognize a problem with a recording. Too much bass, too little treble, too much dynamic range…whatever the problem, this person will work to rebalance things so they work better. In fact, they will probably sound more like the engineer heard them in the mixing studio.

Mr. Make-It-Better

This is the hot shot who knows how to add that little extra pinch of spice, whether it be sparkle or fullness or depth. Mr. Make-It-Better can take a good mix to an even better place.

Ms. Nuts-and-Bolts

This is the nerd. There are no sexy tricks and she doesn't wear a superhero cape, but Ms. Nuts-and-Bolts is the person who will be sure that everything is done and done right. Under her watchful eye, every master that goes out is without technical flaws, and she'll use the least amount of processing to get the best possible result.

In truth, every mastering engineer has all three personalities within him/herself, and knows when to call on each during the process of mastering.

3: MASTERING BASICS

MIXING VS MASTERING

We caution you against doing mixing and mastering in one step—that is, trying to master while simultaneously mixing a multi-track project. When trying to achieve both at once, you're tempted to try to mix, master, arrange, and maybe even re-record within the same session. The separation of recording/mixing and mastering is very important. When mastering, you primarily focus on the overall sound of the mix and improving that, instead of thinking "I wonder how that synth part would sound with a different patch?" If you focus too much of your work on a single instrument in a complex arrangement, you likely will miss the fact that even if you have improved the sound of that one instrument, everything else has been impacted negatively. Get the mix you want, mix down to a stereo file, and then master as a separate last step.

OZONE TIP

For enhancing and finessing the sound of each track in the mixing stage, iZotope offers Alloy 2 (www.izotope.com/alloy), a collection of essential mixing tools.

An essential part of learning to master is to practice by mastering the work of others. It gives you good practice to listen to a wide variety of balances, tones, and dynamic range. Every engineer and producer has their own take on these things. A mastering engineer's job is to try and get the vision of the engineer and producer (and ultimately the artist) to speak as clearly as possible. Once you have some experience experimenting with what sort of changes work or don't work, you can do a better job of stepping back and evaluating your own projects with a slightly more objective ear. However, even seasoned engineers prefer to have someone else master their work as they value the fresh perspectives that outsiders bring.

MASTERING EFFECTS

When mastering, you're typically working with a limited set of specific processors.

- **Compressors, limiters, and expanders** are used to adjust the dynamics of a mix. For adjusting the dynamics of specific frequencies or instruments (such as controlling bass or de-essing vocals) a multi-band dynamic processor might be required. A single band compressor simply applies any changes to the entire range of frequencies in the mix.
- **Equalizers** are used to shape the tonal balance.
- **Reverb** can add an overall sense of depth to the mix, in addition to the reverb that may have been applied to individual tracks.
- **Stereo Imaging** can adjust the perceived width and image of the sound field.

- **Harmonic Exciters** can add an edge or "sparkle" to the mix.
- **Limiters/Maximizers** can increase the overall level of the sound by limiting the peaks to prevent clipping.
- **Dither** provides the ability to convert higher word length recordings (e.g. 24 or 32 bit) to lower bit depths (e.g. 16 bit for CD) while maintaining dynamic range and minimizing quantization distortion.

With all these types of effects, you might wonder where to start. First off, remember, just because you have all these modules doesn't require that you use them all. Only use as many as you need. In truth, there really isn't any single "correct" order for effects when mastering, and you should feel free to experiment.

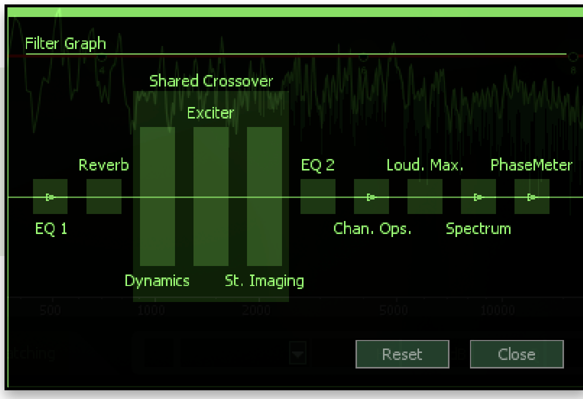
My preferred order usually is:

1. Equalizer
2. Dynamics
3. Post Equalizer
4. [Harmonic Exciter]
5. [Stereo Imaging]
6. [Reverb]
7. Loudness Maximizer

The processors less frequently used are in brackets.

TIP If there is something that comes close to being an iron-clad rule, it is that when you're using the Loudness Maximizer and Dither, they should be placed last in the chain.

TIP For a complete guide on dither, please check out iZotope's dithering guide at www.izotope.com/ozone/guides



OZONE TIP

To change the order in Ozone, click the "Graph" button. This brings up a display of the modules. You can reorder the modules by simply dragging them around.

Note that the location of the meters in the signal chain can also be changed. This allows you to set whether the spectrum is based on the signal going into or coming out of the EQ, for example.

PRESETS

It seems a good moment to talk a little bit about presets. Presets have several uses. The most basic and useful type of preset is one that helps you get up and running right off the bat. If you find that you commonly set up a particular order of effects when you're mastering, having a preset that's configured that way from the get-go is a no-brainer useful idea. Further, if you want a good starting point for achieving a particular effect or task (for instance, "3dB of limiting with a level-matched A/B compare" or "brighter with a high frequency shelf"), a preset for that scenario is a fine thing to have at your fingertips.

Presets can also be valuable as a starting point to educate your ear. There are oodles of presets in Ozone, for example, that are very varied. By trying different presets you can begin to learn what different tools, in different combinations "do" to the sound. Remember, in order to really evaluate them, you need to have the level-matched when you turn them off and on again.

Personally I find presets to be less useful as a starting point for my final processing version. A preset for "Reggae" or "RnB Thump," for instance, might be interesting as they point to an idea about what makes reggae sound like "reggae," but the truth is that a preset can't really take into account what your Reggae track sounds like and therefore it can't possibly know what your Reggae track needs. Maybe your Reggae track has too much bass and the preset is designed to add bass. Perhaps you want your Reggae track to be different from most prototypical Reggae. Giving a quick listen to a track through a complex preset might be interesting, but if I engage a preset I often spend more time turning processing OFF than I would designing a processing setup from the ground up.

Hopefully that helps you understand ways in which presets can be useful and ways in which you need to be very careful engaging them.

GENERAL RECOMMENDATIONS WHILE MASTERING

While you should educate yourself about the function of individual tools in your toolbox, ultimately the tools themselves do not make the sound. They are designed to *help* the sound, so you'll want to decide what sort of help the sound needs. This may sound obvious but just like a good cliché, the obvious truths are often obvious because they are so very true. The fact is that nowadays we have digital signal processing (DSP) tools that are vastly powerful and allow you to change, twist, repair, and contort your sound a million different ways. It is also true that the more involved the processing, the greater the potential for harming the original sound. A multi-band tool will do much more "damage" than a single-band tool. A mid-side process will create problems that a standard stereo processor will not. BE CAREFUL! Before deciding you need the latest whiz-bang feature, figure out what the goal is. Then you can decide which tool is best to use.

The Mastering Mindset

Your thought process might go something like:

Step 1 – Listen: "Hmm, I think I have identified something I would like to change."

Step 2 – Assess: "What tool or technique would be best to make that change?"

Step 3 – Experiment: "Let's try it out."

Step 4 – Evaluate: "OK, I tried it…but did it work?"

Once you decide if your experiment worked, you can determine whether you need to go back to Step 1 or Step 2. Do so as many times as necessary to get to where you are satisfied.

MASTERING QUICK TIPS

1. Have someone else master your mixes for you. In many project studios, the same person is often the performer, producer, mixer, and mastering engineer. If hiring a trusted mastering engineer isn't an option, at least get someone else to listen with you. You could also find someone who will master your mixes if you master theirs. Why? Well, if you have the tendency to add too much bass or not enough top end due to your listening environment, those tendencies will be compounded in the mastering. It's common for the mix engineer to be too close to their own music. You'll focus on some things other listeners won't hear, and you'll miss things that everyone else does hear.

2. Take breaks and listen to other CDs in between sessions. Refresh your ears in terms of what other stuff sounds like. Even seasoned pros, who instinctively know what sound they're working towards, will take a moment to listen to a familiar recording and recalibrate themselves during a session.

3. Listen on other speakers and systems. Burn a CD with a few different tracks and play it on your home stereo system, or drive around and listen to it in your car. Don't obsess over the specific differences, but just remind yourself what other systems sound like.

4. Check how it sounds in mono. This can't be stressed enough. A good ratio between mono (correlated) and stereo (uncorrelated) information is very important in many contexts; broadcast, LP/

vinyl cutting and even MP3 creation. When you listen in mono and important instruments vanish, or if the level drops significantly, you might need to rethink what you are doing.

OZONE TIP

Ozone (and Insight) provide a quick check for this using the Vectorscope in the Stereo Imaging module. Using Ozone you can quickly switch to mono, switch left and right speakers, and flip the polarity of speakers.

5. Monitor at around 85dB SPL (C-weighted). How loud is that? Turn up your speakers until you can still have a conversation with someone who is a meter away without having to strain your voice. That's just about right. When you listen at low to medium volumes, you tend to hear more midrange (where the ear is most sensitive) and less of the lows and highs. This is related to something called the Fletcher-Munson effect, which involves how different frequencies are heard differently depending on the playback volume. So check from time to time how it sounds at different volume levels.

6. When you think you're done, go to bed, and listen again the next morning.

The Tools of the Trade

In the following chapters, I'll briefly step through the standard mastering toolkit with some related thoughts about each tool. This is not meant to be a comprehensive guide, but will hopefully help give you some focused thoughts about the tools and their uses. Over time, you can continue to learn by experimenting; even through failing, then refining your process, you can improve your techniques.

In case it's not totally obvious, I will give the following perspective again—the tools are a means to an end, not an end in and of themselves. We don't use an EQ just to use it. We use it because we think we need it based on what we perceive. In every instance, we always want to do only what we think we need to do to make something better. No less and no more.

4: WHAT IS OZONE?

As we explore the mysterious world of mastering in greater depth, we'll be using iZotope's Ozone to demonstrate key concepts along the way. Ozone is a software plug-in for use within a DAW, encompassing several processing modules in order to provide a complete suite of audio-specific mastering tools. In addition to providing audio processing, Ozone also includes a number of visual meters, useful tools and gauges for evaluating mixes. The tools within the Ozone plugin can be reordered in various ways, and combined with other software and hardware tools to suit both your desires and the needs of your mixes.

Though Ozone is a software product, it combines the best of both the analog and digital domains. Which should you favor? It's entirely subjective, but we'll explore both approaches.

ANALOG MODELING

Given that the analog world is an unpredictable place, it's nearly impossible to use digital 1s and 0s to precisely model all aspects of analog equipment. However, Ozone provides the option to recreate the analog sonic behavior of dynamics processing, equalization and harmonic excitation. Using very high precision, Ozone's analog modeling translates into a great analog-sounding result with all the benefits of the digital world: repeatability, flexible routing, automation of controls, portability, no limits due to analog circuit designs, and freedom from servicing hardware components (take *that*, Mr. Analog!).

So what is this "character" of analog? There have been volumes written on this topic, and we are all still working to refine our understanding of it. In the most general sense, analog processing has certain nonlinear aspects (noise, phase distortions, chaotic unpredictable behavior) that a "purist" might consider "wrong" but in some cases translate to a "musical" sound. Any analog equalizer, for example, applies a small phase shift to the sound.

These types of "imperfections" provide the analog characteristics that some call warmth, thickness, sparkle, or simply an "overall pleasing" sound.

DIGITAL PRECISION

While analog modeling can provide a character or "colorization" of the sound, in some situations precise or "transparent" signal processing is desired. Well-designed low distortion digital processing can help you retain depth, sharpness and detail. For example, you may wish to equalize or notch out a frequency without introducing the phase delay inherent in analog filters as mentioned above. For these applications, plug-ins sometimes provide digital or "linear phase" equalizer modes and low distortion crossovers and filters. As an added bonus, using software-based tools opens up so many creative possibilities, from dreaming up new types of "circuits" to full automation of parameters and incredible portability.

METERS AND DSP

A mastering engineer's hearing needs to be very acute and well developed so he or she can hear a sound and know its frequency, or hear a sound and know when and how compression is working. The ears, though important, are not the only tool needed to evaluate a recording. For most, visual feedback is also very important, and thus mastering engineers also need meters. Each module within Ozone combines both audio processing controls and visual feedback in the form of various meters, traces and spectrograms. When equalizing, you can see a spectrum. When compressing, you can see a histogram or gain trace (a meter highlighting the activity of the gain reduction) of the compression. When listening for the stereo image width, you can watch phase meters.

There is no substitute for using your ears, but think of it like driving a car. When you first start driving, you might spend a lot of time checking the speedometer to get a sense of what's happening. Over time, you develop an instinct and need the speedometer less. But from time to time, we've all looked down and thought "hmmm, I had no idea I was driving that fast." Whether using Ozone or not, whether you're just starting with mastering or have been doing it for years, you can always benefit from the second opinion that a good set of visual displays can provide. These visual displays may also aid in diagnosing specific problems, saving you the precious commodity of time!

OZONE TIP

To that end, iZotope released a suite of meters called **Insight** (www.izotope.com/insight) to add even more detail to the visual feedback about your audio. Insight is also included as part of Ozone 5 Advanced.

5: EQUALIZER

A reasonable starting point when mastering is equalization. While most people understand how equalizers work from a practical perspective and what they can do, it's not always easy to work effectively with one.

WHAT'S THE GOAL OF EQ WHEN MASTERING?

When we're trying to get our mixes to sound good, one thing we're shooting for is a "tonal balance." Any instrument-specific equalization has hopefully been done during the arranging and mixdown stages, so we're just trying to shape the overall sound into something that sounds "natural." This might mean using EQ for both subtle correction and sonic enhancements, but only as needed. Subtlety is the key word here: indeed, the experienced mastering engineer may make noticeable sonic improvements with even slight changes of anywhere between +/- 0.5 to 1.5dB.

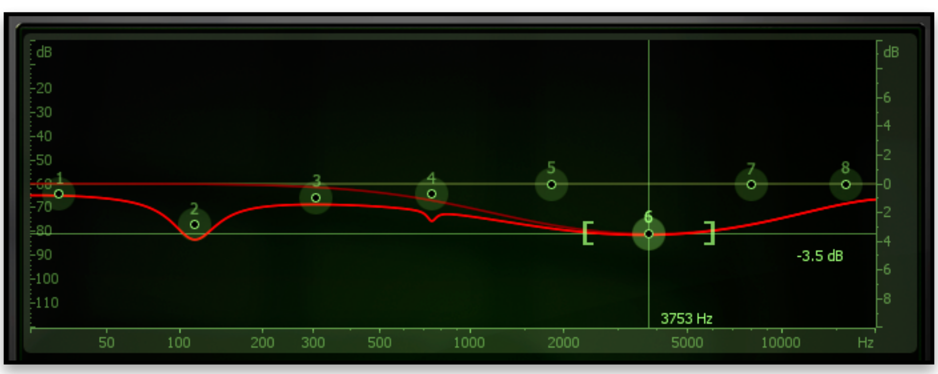
Sometimes that's easier said than done, but there are some general techniques you can use to get a decent tonal balance.

PRINCIPLES OF EQ

There are many different types of equalizers, and they are all meant to boost or cut specific ranges of frequencies. EQs are typically made up of several bands. A band of EQ is a single filter. By combining bands, you can create a nearly-infinite number of equalization shapes.

Parametric equalizers provide the greatest level of control for each band. They allow for independent control of the three variables, amplitude, center frequency, and bandwidth that make up a bell or peaking equalizer.

The picture below shows the equalizer screen in Ozone, but the principles are the same for most parametric EQs. There are 8 sets of arrows, which represent 8 bands of equalization. One band is selected, and has been dragged down to cut the frequencies in the range of 3753 Hz by –3.5 dB. The bright red curve shows the composite or overall effect of all the bands combined. The darker red curve shows the effect of the single band that's selected.



Each band of a parametric equalizer typically has three controls:

Frequency

The center frequency dictates where the center of the band is placed.

Q and/or Bandwidth

Q represents the width of the band, or what range of frequencies will be affected by adjustments to the band. A band with a high Q (see Figure 1) will affect a narrow band of frequencies, where a band with a low Q (see Figure 2) will affect a broad range of frequencies.

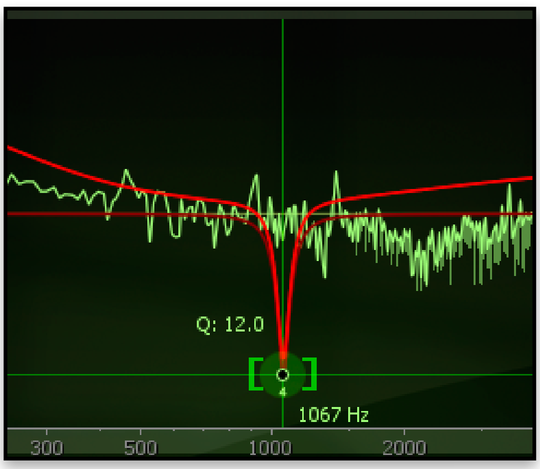


Figure 1: a narrow filter (Q=12)

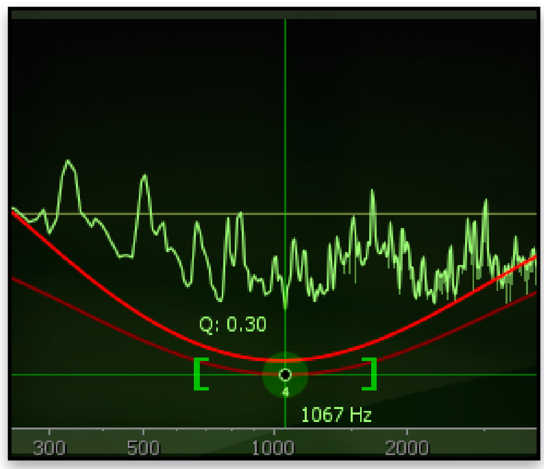


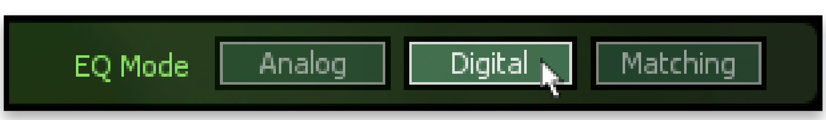
Figure 2: a broad filter (Q=0.30)

Gain

This determines how much each band boosts (turns up) or cuts (turns down) the sound at its center frequency.

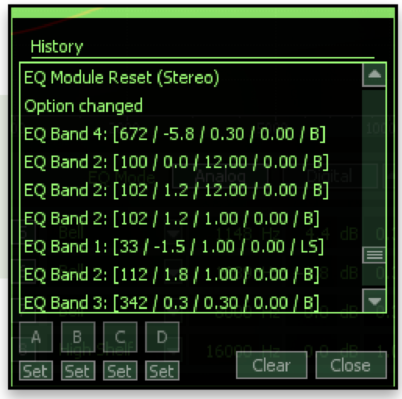
Should I Use Digital or Analog EQ ?

Analog filters, as mentioned before, impart a certain character or "color" to the sound. If your goal is less of the "time-domain smear" that comes along with analog-style processing, you can use a digital linear phase EQ, as demonstrated below.



The selection is a matter of taste, although in general (or in my opinion) analog/analog-modeled filters provide an excellent sound when applying slight boosts or cuts, while the "transparency" of digital linear phase filters is useful when applying deep or narrow "surgical" cuts.

Over time, as you become more familiar with the sound of different EQ filters, you'll find the decision easier to make—though, to some degree, there is always that element of experimenting, listening, and verifying that your chosen EQ type and settings are making genuine improvements to the audio in question.



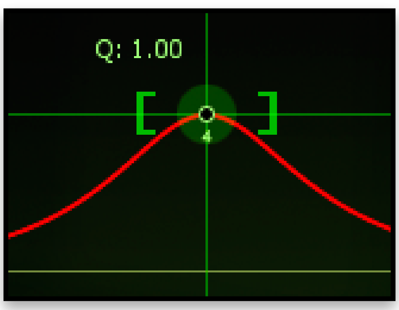
OZONE TIP

For a better idea of what sounds best for any specific scenario, use the unlimited Undo History to quickly audition between various EQ filter types on the fly.

EQ Shapes

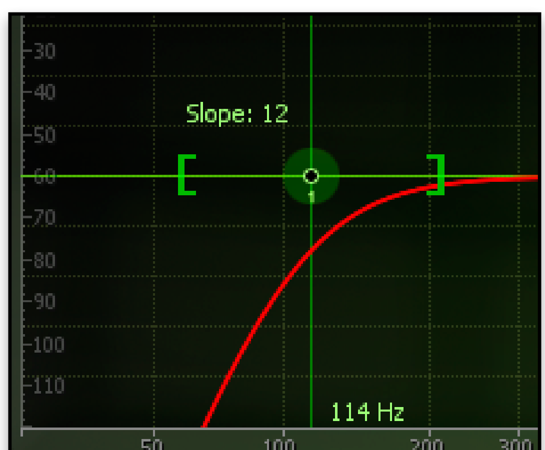
Bell EQ

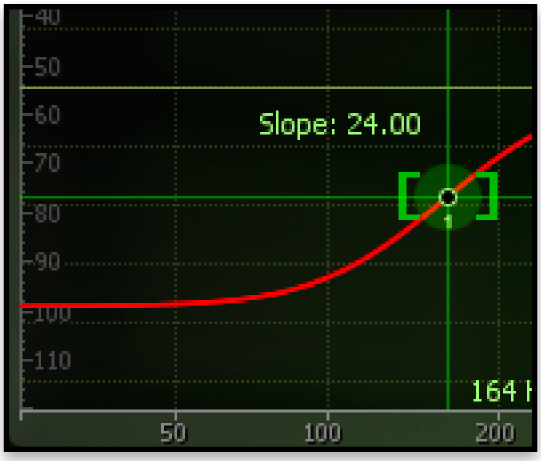
A bell filter has a width (Q) as well as a gain. The gain can be positive or negative, to either boost or cut the specified range of frequencies within the bell.



Lowpass and Highpass Filters

Unlike a bell filter, lowpass and highpass filters only have one "side" to them. You specify the point where you want to start attenuating frequencies and any frequencies below that point (for a highpass filter) or above that point (for a lowpass filter) are rolled off more or less steeply. The Slope setting specifies the grade of the filter's roll-off, with a lower slope resulting in more gradual roll-off of frequencies.



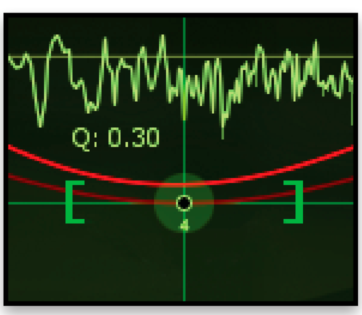


Low Shelf and High Shelf EQ

Like lowpass/highpass filters, these filters also are "one sided." Shelf filters, however, don't drop off indefinitely. Instead, they resemble, well, a shelf. In this case, the horizontal handles provide a Slope control which specifies how tall the shelf should be, or how much cut should be applied before leveling off to a constant (horizontal) line.

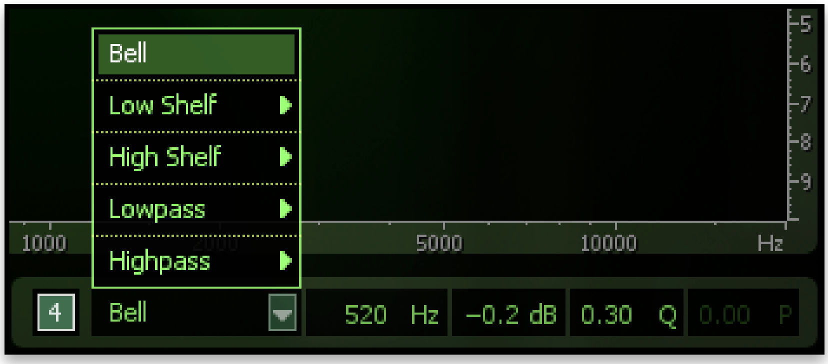
TIP Certain analog shelving filters exhibit additional behaviors that can sound fantastic in the right context, such as the complementary frequency dip of the Pultec EQ, a characteristic that is also available in Ozone (look for the "vintage" shelving EQ type).

USING THE OZONE EQUALIZER



Ozone includes two parametric equalizers presented in a graphical way, which is referred to as a paragraphic equalizer. Each Equalizer module has 8 adjustable filter bands which can be used to boost or cut frequencies. To adjust the gain of a band, you grab the center and move up or down. To adjust the frequency, you drag left or right. To adjust the Q or width of a band, you can grab the side handles of the band and drag them apart or click on a filter and scroll the mouse wheel up (to decrease the Q and widen the band) or down (to increase the Q and narrow the band).

Any of the eight filters in Ozone can be configured to be one of several types of bell (also referred to as a peaking filter), lowpass, highpass, low shelf or high shelf filters. There are three main ways you can specify the shape of a filter. First, you can choose the filter shape by selecting a node and opening its individual shape selection pop-up menu ("mini info-panel"), just below the EQ screen.



Second, and usually the easiest way, is to simply right-click a node and a filter shape selection menu will pop up. Finally, you can select the shape of a filter by clicking on the button, opening the EQ Bands tab and selecting a different shape for the filter from the table drop-down menus.

Controls for Adjusting EQ Bands

In addition to basic mouse support, Ozone supports the following controls for adjusting EQ bands:

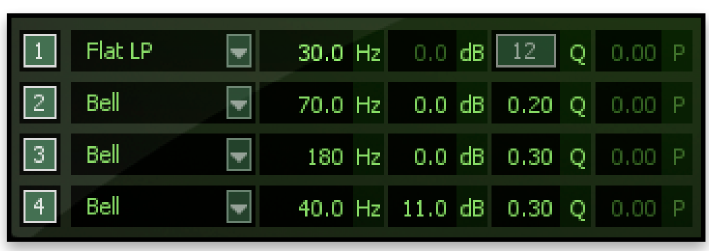
1. You can use the arrow keys to adjust a band up/down or left/right. If you hold down the Shift key when using the arrow keys the adjustment is accelerated.

2. You can adjust the Q of a band by using the wheel of a wheel mouse or the PgUp/PgDn keys.

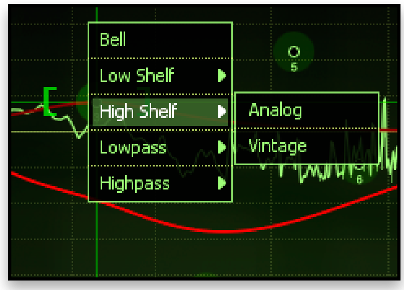
3. You can select multiple bands by holding down the Ctrl key and clicking multiple bands. To adjust them as a group, drag one of the selected bands and the rest will move with appropriate relative motion (or use arrow keys to move the entire group). This is useful if you have an overall shape that you like but want to raise or lower the gain of the entire curve.

4. If you hold down the Shift key and drag an EQ band, the EQ band will be "locked" in the direction that you're dragging. If you just want to change the gain without affecting the frequency (or vice versa), just hold the Shift key while you drag.

5. If you'd rather use numbers as opposed to visual EQ bands, opening the EQ Bands tab gives you a table view of the EQ band settings. You can enter values for the EQ bands directly in this table, or simply position the cursor over a value and change it by turning the wheel of your mouse or dragging the values up or down. You can also disable bands with this table by clicking on the square box to the left of a band.



6. You can select the shape of a filter by right-clicking on the EQ filter you want to change directly in the EQ graph.



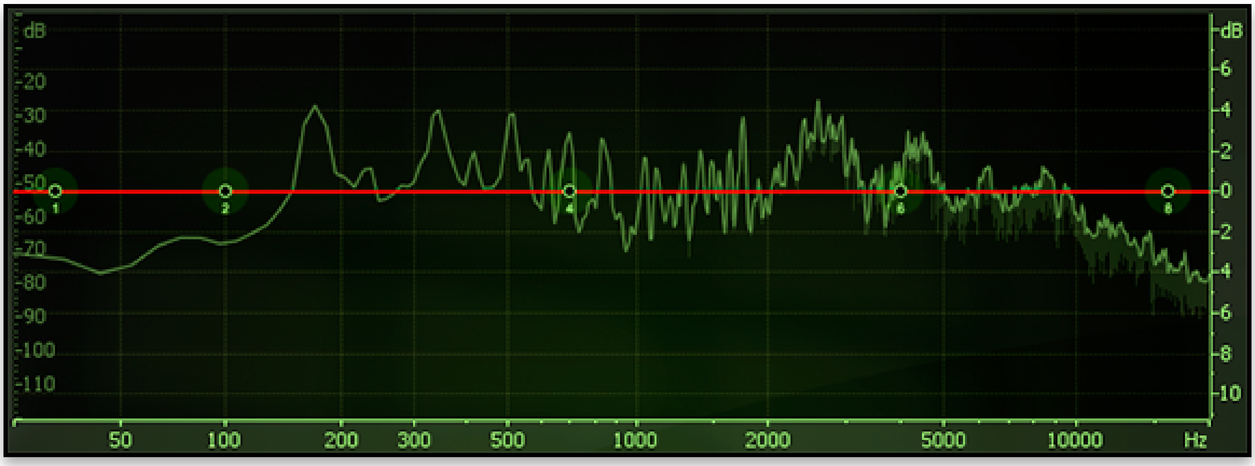
OZONE TIP

If you hold down the Alt key and click on the spectrum, you reveal an "audio magnifying glass" that lets you hear only the frequencies that are under the mouse cursor, without affecting your actual EQ settings. This is useful for pinpointing the location of a frequency in the mix without messing up your actual EQ bands. Releasing the mouse button returns the sound to the actual EQ. You can set the width of this filter in the Options dialog by adjusting the "Alt-solo filter Q." Double-click in the spectrum area to add an EQ band. It's a useful workflow to alt-solo and find a problem frequency, then double click exactly where the mouse is to add a new EQ band at that frequency. Then you can hold shift to drag that band down and cut those frequencies (see next section).

EQ'ing with Visual Feedback

The key to setting the tonal balance of a mix with an EQ is developing an ear for what frequencies correspond to what you're hearing. A spectrum analyzer is helpful for allowing you to confirm and assign numerical values to what you hear. The following information will help you understand the options that the spectrum analyzer gives you, but be wary. While it is very good at showing you the maximum energy along the spectrum over time, it tells you nothing about the mix, the sound of individual instruments, the style and internal dynamics. Don't become obsessed with the analyzer. However, if you have a boomy kick or significant sibilance in a mix, it's usually pretty easy to pick it out with a spectral analyzer and address it.

The spectrum analyzer from Ozone is shown below, although others provide similar views and options. The green line represents the spectrum or FFT, calculated in real time, ranging from 20 Hz to 20 kHz, the range of human hearing.

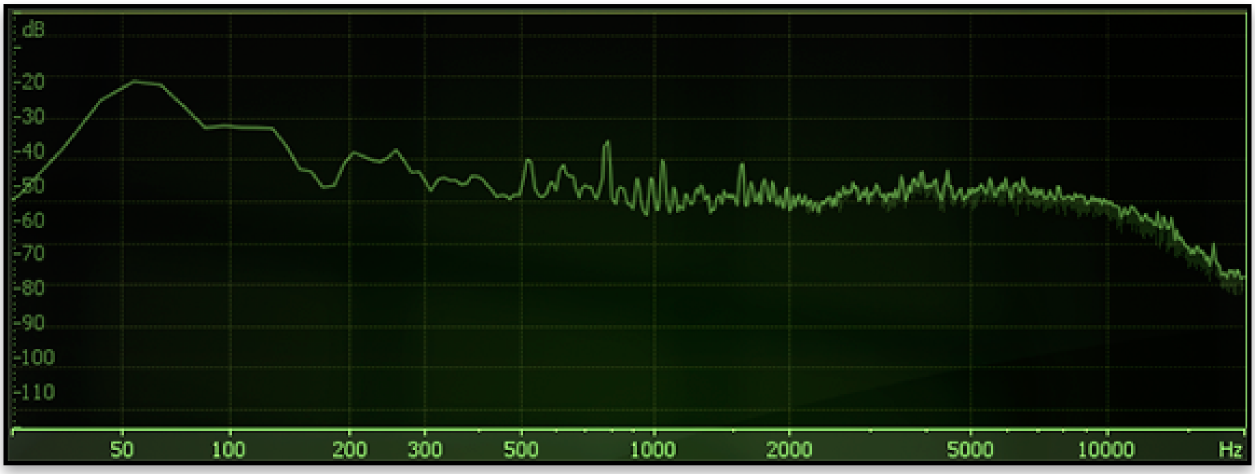


Peaks along the spectrum represent dominant frequencies. In the case of the song above, you can see a slight dip in frequencies between 100 and 300 Hz, which could be compensated for by using mid-low frequency EQ or mid-low compression.

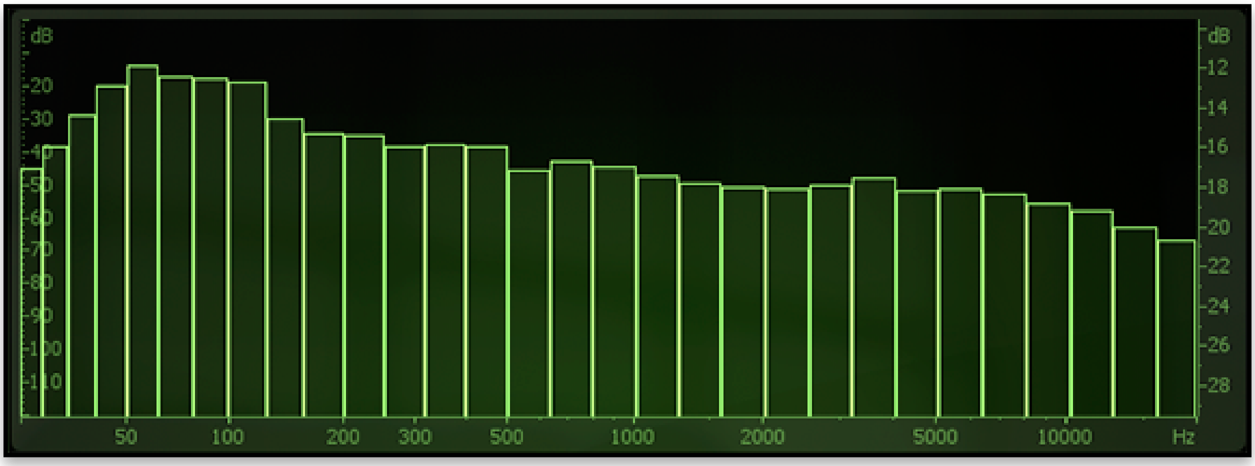
Spectrum Type

Ozone allows you to select between Linear, 1/3 Octave, Critical Bands, and Full Octave spectrums.

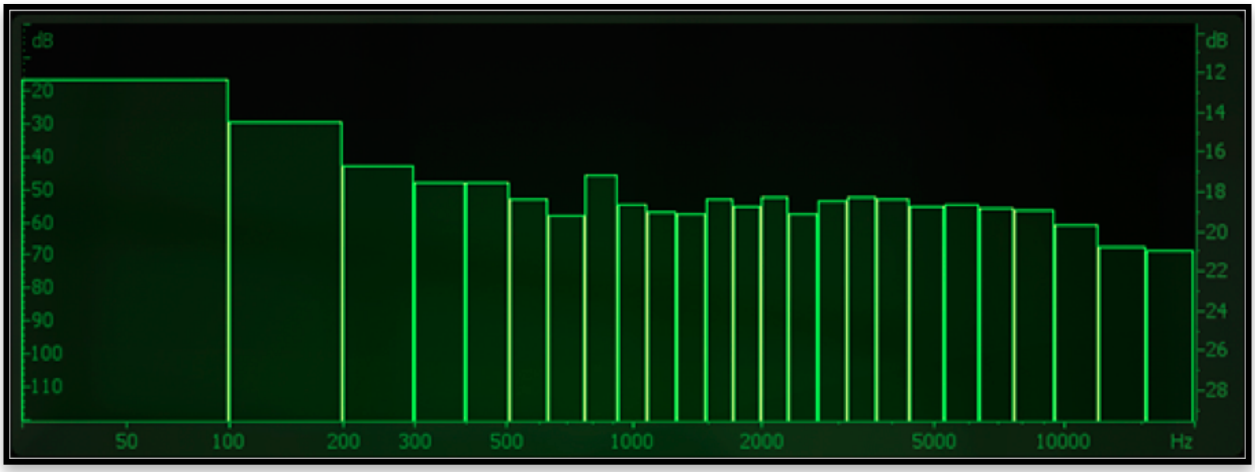
The **Linear** spectrum is a continuous line connecting the calculated points of the spectrum, as shown below.



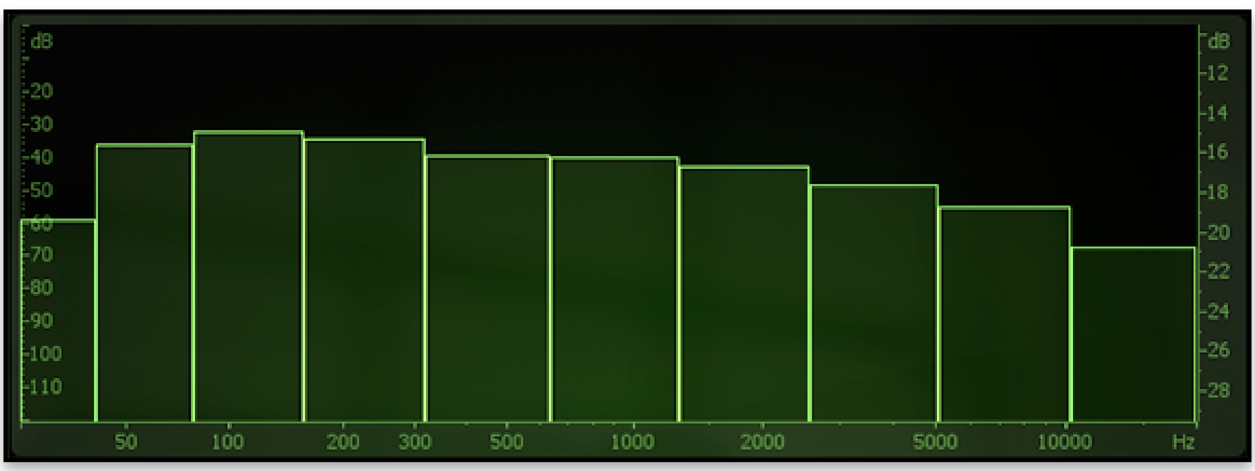
A **1/3 Octave** display splits the spectrum into bars with a width of 1/3 of an octave as shown below. Although the spectrum is split into discrete bands, this option can provide excellent resolution at lower frequencies.



The **Critical Bands** option splits the spectrum into bands that correspond to how we hear or, more specifically, how we differentiate between sounds of different frequencies. Each band represents sounds that are considered "similar" in frequency. A critical band representation is shown below.



The **Full Octave** option splits the spectrum into bars with widths of full octaves covering the entire range of the frequency spectrum.



- **Peak hold:** Allows you to show and hold the peaks in the spectrum. (note that in Ozone you can reset the peak hold at any time by clicking on the spectrum).
- **Average time:** If you're concerned with peaks or short frequencies you can run the spectrum real time mode. For comparing mixes and visualizing the overall tonal balance, Ozone also provides an averaging mode. Instead of overwriting the display of old samples with new samples, Average mode averages new samples into the prior samples to provide a running average of the tonal balance. You can reset the average at any time by clicking on the spectrum.

- **FFT Size:** Without getting into the math, the higher the FFT size, the greater frequency resolution. An FFT size of 4096 is usually a good choice, although you can go higher if you want better resolution, especially for focusing in on lower frequencies.

- **Overlap and Window:** These are more advanced options that determine how the window of audio is selected and transformed into a frequency representation. In general, an Overlap of 50% and a Hann window will give good results.

OZONE TIP

You can turn off the spectrum display from the Ozone main options dialog to conserve CPU or to minimize visual distraction.

Snapshots

Spectrum snapshots are powerful tools for comparing the tonal balance of your mix to other songs. In Ozone, these snapshots can be accessed by clicking on the Snapshots tab.

You have access to eight Snapshots, marked by different colors and able to be renamed to whatever you like. Clicking on "Start Capture" takes a snapshot of the spectrum at that instant in time. You can show individual snapshots by clicking the "Show" checkbox next to each Snapshot name.

Choose the amount of audio you want to average in building your snapshot in the "Snapshot Time" drop-down menu, click "Start Capture" and wait until enough audio is captured to build the snapshot. When ready, click "Stop Capture" and your snapshot will be added to the list of snapshots and displayed on the spectrogram automatically.

GENERAL EQ TIPS

So you're ready to EQ. Now what?

Listen and try to identify any problems that you hear. Start with the midrange (vocals, guitar, midrange keyboard, etc.) as this will typically represent the heart and soul of the song. Does it sound too "muddy"? Too nasal? Too harsh? Compare it to another mix, perhaps a commercial CD.

Try to describe to yourself what the difference is between the two mixes around the midrange.

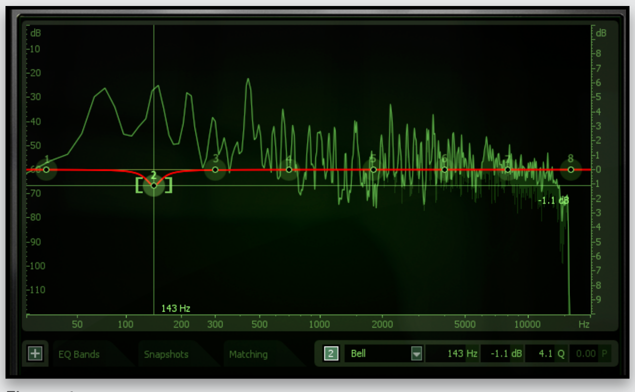


Figure 1

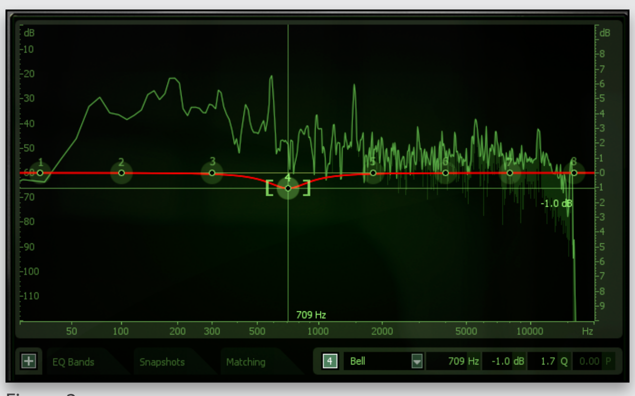


Figure 2

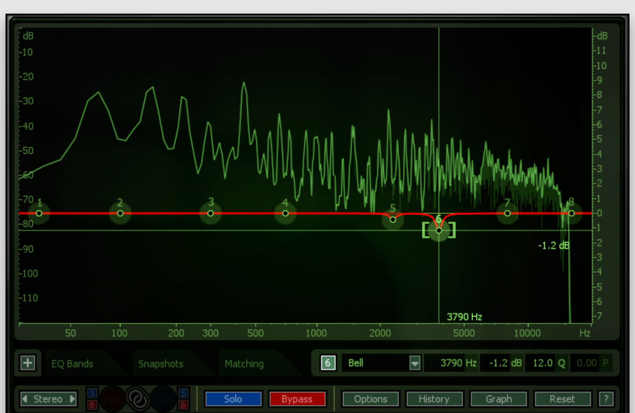


Figure 3

MASTERING QUICK TIPS

Too muddy?

Try cutting between 100 to 300 Hz (Bands 2 and 3 are set at these frequencies by default in Ozone—try cutting the gain in this region a few dB using these filters). [see figure 1]

Too nasal-sounding?

Try cutting between 250 to 1000 Hz (Band 4 in Ozone is set by default at 700 Hz for this purpose). [see figure 2]

Too harsh-sounding?

This can be caused by frequencies in the range of 2000 to 3500 Hz. Try cutting this range a few dB. Hopefully, using a band or two in these regions will give you a better sounding midrange. Remember that you can use the Alt-click feature to focus just on specific ranges and highlight what you're hearing. A common technique is to start by boosting a band to dial in a region of the spectrum that contains too much energy, and then cutting it once you've centered on the problem area. You'll get the most natural sound using relatively narrow bands when cutting, but when boosting, wide bands generally sound more 'musical' (a Q less than 1.0). [see figure 3]

Once you contain the problematic areas in the midrange you can move on to addressing the broader tonal issues. Does the sound need to be brighter? Have more or less bass? Try making a few adjustments, then step back and shift your attention back to the midrange…then back again. Given the way we experience the complex signal that makes up music, there's really no way to simplify the process so that you always do the same thing every time, or use the same number of steps every time. Each piece of music presents its own challenges. It usually takes some time before you come to a satisfactory result, but the following guidelines may help.

MASTERING QUICK TIPS

1. Try cutting bands instead of boosting them.

2. Cutting or boosting more than 2-4 dB means you probably have a problem that you can't fix from the stereo master. Go back to the multi-track mixing step.

3. Use as few bands as possible.

4. Use gentle slopes for boosting (wide bandwidth, low Q) and narrow bands for cutting.

5. Shelf or highpass filters below 30 Hz can get rid of low frequency rumble and noise, but it comes with a price. Listen carefully to be sure the rest of the audio doesn't sound worse.

6. Use your ears and your eyes. Compare to other mixes using both senses.

If you find yourself using too narrow of a notch filter, or too much gain, you may be trying to fix something that EQ on a stereo mix can't fix. Go back to the individual tracks and try to isolate the problem that way. Note also that the wider the band, in general, the less gain you need to apply.

In addition, your ears quickly get used to EQ changes. You may find yourself boosting more than necessary to hear the difference. Use the History window (click on the History button) to go back and audition settings prior to making changes. Comparing the difference before and after a series of subtle EQ changes can help prevent you from overdoing boosts or cuts.

Remember to check your ABCD's!

A is reference music that you think sounds good in your room.

B is the original mix – are you making it better than the original?

C is the new version of your track – is there anything you are doing that is revealing something unattractive?

D is any other tracks/songs that will be part of the same collection. Does your new version of the track fit with the others?

If you are initially struggling to make informed and intelligent EQ choices, don't worry! Mastering is sometimes like a big game of Sudoku…figuring out how to make all the pieces work together. Time and practice are the answer.

6: DYNAMICS

Dynamics in mastering isn't just about making things "louder" and more "competitive." The primary aim with which we began this guide was to make things sounds at least as good, if not better, than they did in the mix. So where does dynamics processing fit in?

WHAT'S THE GOAL OF DYNAMICS PROCESSING WHEN MASTERING?

A consistent listening experience is one of the desirable effects of a good master. In some cases, such as a classical recording, a wide dynamic range is expected and enjoyed, but in many other cases, the listener does not want to have to constantly reach for the volume control between the verse and a chorus of a song, or even between songs on an album.

Dynamics processing can help reduce or expand the dynamic range as needed, which helps empower the listener to enjoy the recording rather than feeling the need to adjust it. It can also provide additional sonic enhancements by transparently highlighting certain frequency elements or instruments within a mix…or the reverse, "smearing" the transients for a more "gluey," "tighter" sound.

PRINCIPLES OF DYNAMICS PROCESSING

Mastering the dynamics of a mix using compressors, limiters, and expanders is probably the most challenging step of the process, but the one that can make the most difference between a "basement tape" and a commercial-sounding mix. Taking the time to get to grips with dynamics processing can be well worth the effort.

There are a few things that make mastering dynamics challenging:

1. The effect is subtle, at least if done correctly. It's not something you clearly hear, like a flanger or reverb or so forth, but instead something that changes the character of the mix. If you think about it, compression removes something (dynamic range) and so what you will hear is the *absence* of something, if that makes sense.

2. A compressor is not necessarily working all the time. Since it changes in response to the dynamics in the music, you can't listen for one specific effect. Level histograms and compression meters (such as those provided in Ozone) can be invaluable for referencing when the compression is occurring, and by how much.

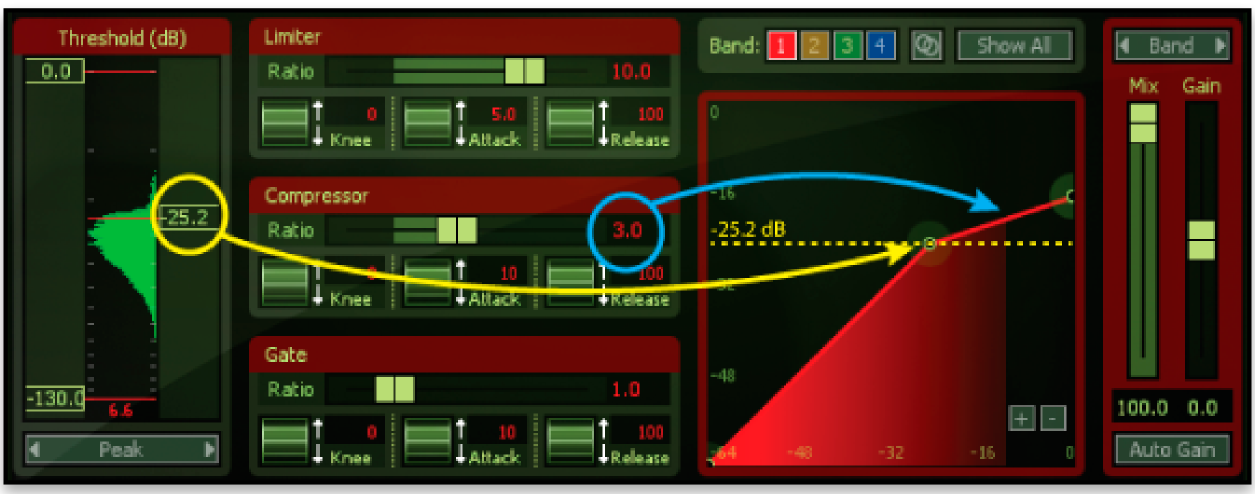
3. Not all compressors are created equal. While the concept is simple enough (restrain the volume when it crosses a threshold), the design and implementation (and therefore the quality) of compressors varies considerably. Applying a quality compressor correctly, however, can smooth the

peaks (and valleys) in your mix and make it sound fuller, smoother or allow you to increase the average level (if that's the desired goal).

USING THE OZONE DYNAMICS MODULE

Ozone includes a multiband, multi-function dynamics processor. Before you jump into the multiple dimensions provided by this module, I recommend you always start with just the simplest case: a single band compressor. An analogy often used for describing compressors is that of a mixing engineer with his hand on the overall output gain while watching the level meter. When the level exceeds a certain point (the Threshold in compressor terms), the engineer starts turning down the level. How much they turn down the level as the output gain exceeds the threshold is called the "ratio." Higher ratios mean that the engineer (or compressor) turns down the volume control more aggressively when the level is above the threshold to bring the output level back down closer to the threshold point. With a ratio of 3:1, if the output level exceeds the threshold by 3 dB the engineer turns down the output level so the net output is only 1 dB above the threshold. As a result, the signal will exceed the threshold level, but not by as much as if there wasn't any compression.

The illustration below will help, and will also introduce the dynamics meters provided in Ozone.

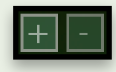


In the screenshot above, the compressor is set with a Threshold of –25.2 dB, meaning that when the signal exceeds –25.2 dB, the compressor will start compressing. The yellow arrow points to this spot on the compression curve.

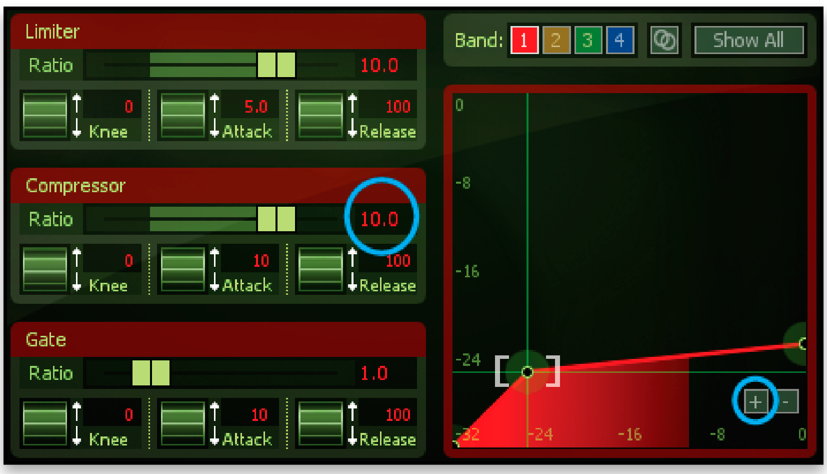
The Ratio is set to 3.0, meaning 3:1. The blue arrow points to the segment of the compression curve affected by the ratio. Everything above the Threshold point is sloped a little less, specifically with a slope of 3:1.

The compression curve therefore gives a visual depiction of the compressor setting. The horizontal line (or x-axis) represents the input signal. The vertical line (or y-axis) represents the output gain. The line or line segments in the graph show what happens to the output level at each input level. So with our settings above, anything above –25.2 dB will start to be compressed.

OZONE TIP

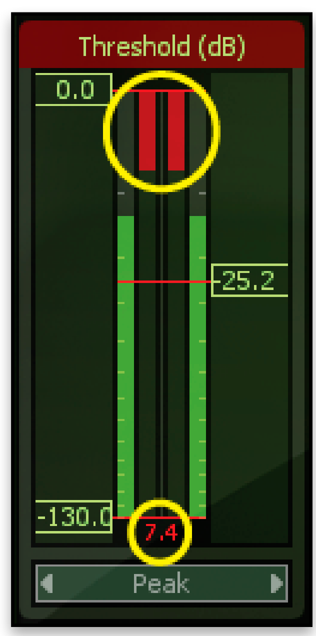
Note that in Ozone you can zoom in or out on the compression curve by clicking on the buttons  or by holding down the Ctrl key and clicking (left-click to zoom in or right-click to zoom out).

So let's turn up the ratio of the compressor and zoom in so we can see more clearly what's happening:



Now the ratio is 10:1. If the input signal exceeds –25.2 dB by 10 dB, the output will only go up 1 dB. The compression curve is much flatter above 25.2 now, indicating that the output (vertical axis) is not going to go up very much even as the input level (horizontal axis) goes up.

Dynamics Meters



Most compressors indicate compression with a reading in dB of how much the signal is being turned down by the compressor. For example, in the picture to the left we know that the signal is being compressed (turned down) 7.4 dB by the compression. This meter is not shown by default, however. We can switch to it from the default Level Histogram (shown below) by going to Dynamics Options (click "Options" or right-click a Dynamics control and select "Dynamics Options") and choosing the "Gain Reduction" setting for the Threshold meter. As the signal is compressed (i.e. it exceeds the threshold), this meter (a "reduction meter") pushes down to show how much signal is being "taken off the top" in decibels. The red number below the meter shows the exact amount being reduced.

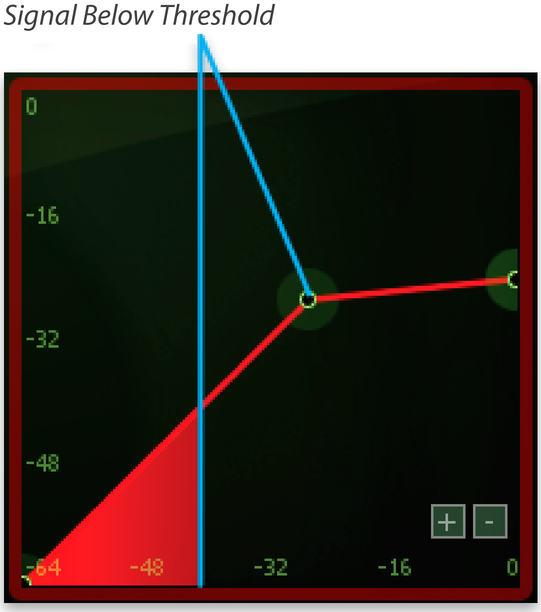
While this type of meter is useful to show what just happened (how much the compressor just compressed), it's not as useful by itself to understand what's happening overall in your mix. Setting the threshold of a compressor involves understanding the history of levels in your mix, so you know where the peaks and valleys are as a whole, not just at any single point in time.

For this reason, Ozone combines a Level Histogram –a meter that shows the history of levels (shown by default in the Dynamics module) –with a reduction meter and a real-time compression curve that shows what's happening at that moment. The Level Histogram shows you where to set the Threshold, the compression curve tips you off to when the compressor is compressing, and the reduction meter shows how much the signal is being compressed. The amount of current reduction is again shown in red below the meter.

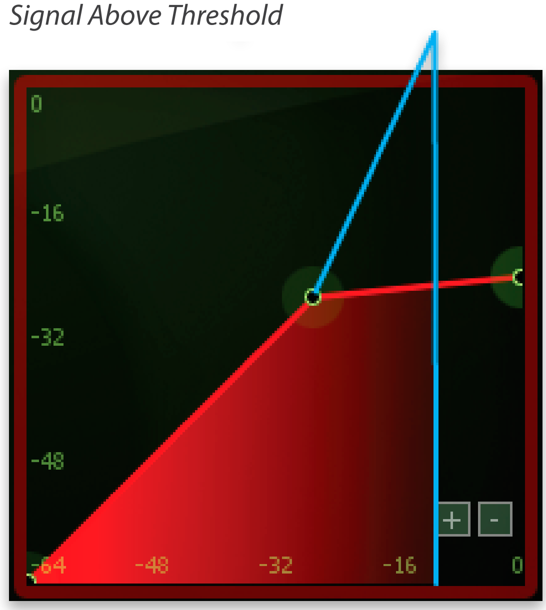
The Level Histogram is shown here. Think of it as a level meter with a memory. As the level changes, it displays a history of where the level has been by showing wider lines for levels that occur more frequently. In the picture to the right, we can see that there is a lot of level around –30 dB, and less around –40 to –60 dB. There is a little bit above the region indicated by the top red circle, and this is our target for compression.

So we set the Threshold at that point. Anything above that point is going to be compressed. Don't worry about dBs and numbers, you can just use your eyes (and ears) to set the point. So how do you know when the signal is being compressed? By using the compression curve meter. In the picture below and to the left, the signal has not crossed the threshold point, so no compression is happening. In the picture below and to the right, the signal has crossed the threshold and is being compressed.

Signal Below Threshold

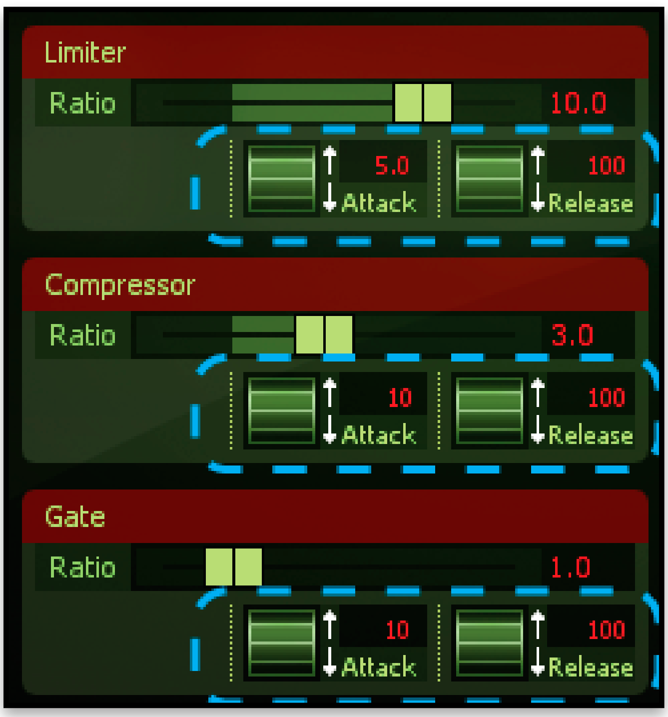


Signal Above Threshold



Simple enough, right? You probably hear all the time that that you can make your mix louder with compressors. A compressor by itself is turning down the level by compressing peaks. But the side benefit is that you can turn up the signal as a whole without overloading, since the difference between the loudest sounds and the softest sounds has been reduced. You do this by turning up the "make-up" gain control of the compressor, which simply adjusts the level of the signal after compression has occurred. As you adjust the gain, the compression curve increases, indicating that the output level (the vertical axis) is now higher.

Attack and Release



The final two settings related to a simple compressor are the Attack and Release controls. You can set these controls by adjusting the Attack/Release rolling dials or by double-clicking the red numbers and manually entering in your desired settings.

Going back to our mixing engineer analogy, these relate to how long the engineer waits (the Attack) to turn down the volume after it exceeds the Threshold, and how long to wait (the Release time) before turning it back up after it drops back below the Threshold.

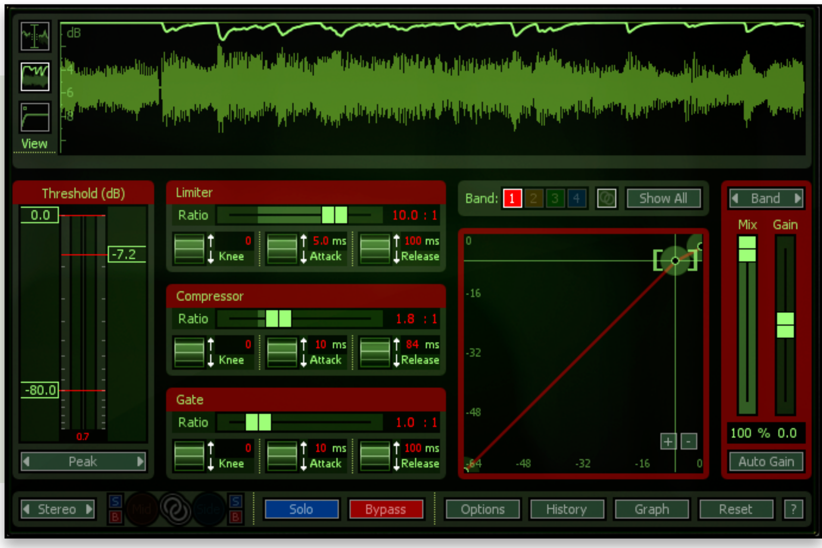
So how do you set these values? Unfortunately there is no simple answer as it depends very much on the sound you are shooting for. Looking at the attack time first, a faster attack will respond more quickly to transients or short peaks in the sound. If you want to soften the attack of a drum, you'll want to set this fast. But maybe you want the "pop" to go through, in which case you'd set it slower. As a rule of thumb, start with the attacks around 20-30 milliseconds. Bring them down to soften the attacks of the instruments, or up to let more of the transients through.

The other thing to keep in mind regarding attack is that too fast of an attack time (and release) can cause distortion (especially in low frequency signals) as the compressor tries to quickly adjust the level. Low frequency signals have long periods (i.e. the length of time it takes to cycle through a fundamental tone), so compression that is adjusting the volume during a cycle can sound very unnatural.

Turning to the release time, this sets how long it takes for the compressor to "let go" and let the level return to normal. As a starting point, try 250 ms, although there is no rule here.

OZONE TIP

You can use the Gain Trace Meter in the top of the Dynamics window to see in real-time what dynamics processing is being applied to your waveform. If performing compression, you may use this trace to check whether the gain line recovers in time for the next transient… if not then you may need to shorten your release time.

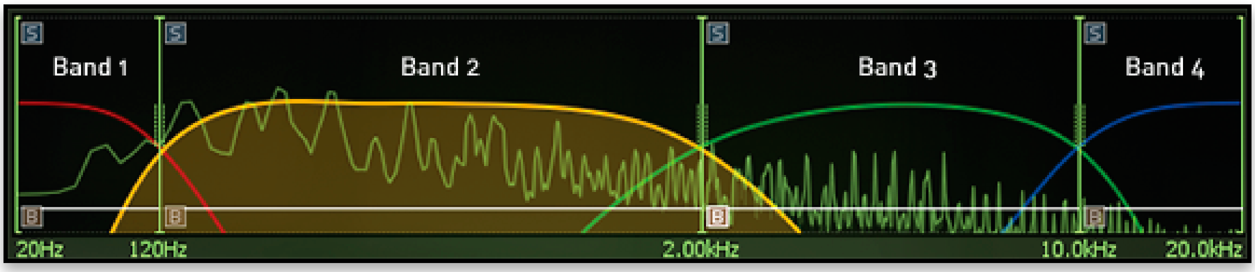


What's important is that you understand the release time concept and implications. Too fast a release time will cause either distortion or a "pumping" sound, since the compressor is releasing and letting the output signal return to normal too quickly. A slower release time will let the level gradually return to its normal unprocessed value. On the other hand, a slow release time will cause the compressor to keep compressing even after loud peaks have passed, and the softer levels that follow the peaks will be unnecessarily compressed.

The power that comes from a compressor is that you can use it to gently compress the middle or average level of the mix. A limiter is suited to taking care of the peaks with a high ratio, you can add "glue" and body to the mix with a low ratio (1.1 to 2.0). Pull the threshold for the compressor down at point where you see 1-3 dB of gain reduction as a good starting point.

Multiband Dynamics

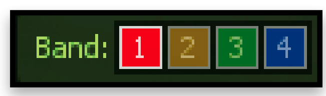
A single band compressor (or combination Limiter/Compressor/Expander) applies dynamics processing to the entire mix, i.e. the entire range of frequencies. Things get more complex when you consider the possibilities of applying dynamics processing separately to individual bands or ranges of frequencies. If you need to, check out our Appendix for more general information on multiband processing. For now, we'll quickly summarize the two main multiband operations in Ozone.



1. The mix is divided into up to four frequency bands. You can set the cutoffs of these bands using the vertical lines or handles on the spectrum in the multi-band modules.

2. You can click on a checkbox in a band to solo (S) the output of that band or bypass (B) Ozone processing for that band. You can Alt-click either checkbox to solo or bypass all modules except the clicked one.

When using Dynamics, clicking on a band displays a set of dynamics controls that are specific to that band. Clicking on the Band Selection button corresponding to the band number you want to control also opens these same band-specific controls. The colors of the control boxes will change depending on which band is selected.



You now have independent control over four separate frequency bands of processing. Click on a band, and you can set the Compressor differently for each band.

OZONE TIP

You can copy and paste settings across bands. Right-click on the mini-spectrum and select "Copy." Right-click on another band, select "Paste" from the same menu, and the settings from one band will be copied over to the other band.

If you're looking for total control of every band with everything in one place, try clicking the "Show All" button. This reveals a window that displays the ratio, attack and release for the Limiter, Compressor, and Gate for each individual band in one screen, as shown below:

Applying a multi-band compressor follows the same concepts as a single band or full range compressor. The difference is that you can apply compression to specific bands. With that in mind, what are the benefits?

The main advantages of a multiband dynamics tool are:

1. You can set different attack times for different ranges of frequencies. A low frequency signal, such as the fundamental of a bass instrument, may take tens of milliseconds to complete one cycle, whereas a high frequency signal like the top end of a hi-hat sound might only take 3-4 milliseconds. With multiband processing, you can tailor the compression to control each range of frequencies to taste. Ideally, then you won't hear the compressor eat up too much bass energy, or not even touch the hi-hat.

2. You can adjust AND control the amount of energy coming out of the compressor for each band. For example, if you need more low end, but don't want the low end to get TOO loud, you can compress and then boost. While an EQ is much better suited to being a tonal control, a multiband sometimes gives just the tool you need.

GENERAL DYNAMICS PROCESSING TIPS

Here's a sequence of steps that you could follow to start using a compressor effectively.

1. Set your ratio. Depending on what you're trying to compress, here are some starting points to try.

   Full mix: Try 1.1 to 2.0 ratio values. It is possible that certain pieces of music, such as club tracks, might benefit from higher ratios.

   Bass, kick: Try 3.0 to 5.0 values. Depending on the sound you're shooting for, you can even go all the way to 10.

   Vocals: Try 2.0 to 3.0. Of course, like everything else in this guide, these are rough suggestions. Your mix, your taste, or your desired effect can radically change where you set these.

2. Bring down your Threshold until it's just above the average level of the mix, which you can see using the level histogram. Aim for no more than 2-3dB of compression at the maximum.

3. Turn up the output gain as you see fit to boost the compressed signal.

4. Experiment with attack and release timings, as there's no good single tip. Remember that shorter attacks will level off more of the transients, but possibly cause distortion. With this in mind, you could shoot for the lowest attack possible before hearing any artifacts.

OZONE TIP

If you're just trying to increase the level of a mix a little without changing the internal dynamic structure, you could use a limiter like the Loudness Maximizer instead. Info on that processor is coming next!

7: LOUDNESS MAXIMIZER (LIMITING)

One of the most common complaints from "project studio" artists is that mixes don't sound loud enough. When you burn a CD of your mix and put it in your CD changer, you may have noticed that your mix just can't compete in rotation with the commercial CDs you have. You turn it up, and it just overloads. What's going on here?

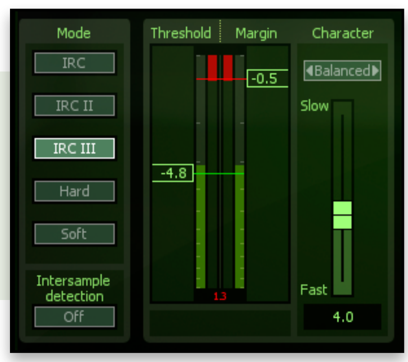
WHAT'S THE GOAL OF LOUDNESS MAXIMIZING WHEN MASTERING?

Unfortunately, recent commercial CDs have been more focused on how a recording's level hits a meter rather than focusing on the sound of a recording. As a result, the overall level of CDs in recent years has reached an extreme. Most artists want their recording to be "competitive" with others in a similar genre, and so it falls to the mastering engineer to decide what that means and how to best get them to that goal.

In order to reach that goal, we can use a combination of compression and limiting (and EQ). Using tools like the Loudness Maximizer to perform limiting is not solely about making a recording "louder," though that is a consideration. Judicious use of a limiter can also enhance the perceived presence and impact of a track. You can even transparently enhance the stereo image by using stereo delinkable limiters (though be careful with these, because it's easy to go too far). Or, you might eschew the latest advances in transparent, crisp limiting technology and deliberately choose a soft and "smeary" analog-style limiter. It all depends on what's appropriate for the particular track in question.

OZONE TIP

Ozone, as it happens, offers five unique limiting options. IRC I, II and III are transparent, psycho-acoustically advanced algorithms, Hard is a brickwall style algorithm, and Soft is more gentle and "smeary." For more info, read on…



PRINCIPLES OF LIMITING

Most sound editors have a Normalize function. The Normalize function analyzes your entire mix, finds the highest peak, and adjusts the gain of the entire mix so that the highest peak in the mix is at 0 dBFS (the verge of clipping) or a specified target level. The rest of the music is then adjusted in level by the same amount. However, all this does is makes the single highest peak on the verge of clipping.

The principle behind a limiter is that you can limit the peaks at the Threshold and then bring up the rest of the mix. The bulk of the mix can be brought up since the peaks are cut down, so nothing overloads 0 dBfs.

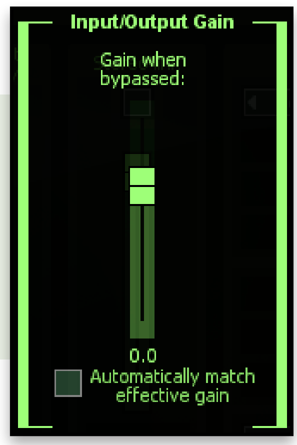
A tiny bit of limiting is almost unnoticeable. In fact if you were to limit or clip a single sample, it is beyond our perception to notice that at all.

USING THE OZONE LOUDNESS MAXIMIZER

Using a limiter is very simple. It usually operates on the entire spectrum. In Ozone it has three sliders, one mode selector and one option.

Threshold

The threshold sets the level at which limiting begins. As you move the Threshold slider down, you are limiting more of the mix. What you will notice as you do that is that the sound appears to get louder. How is that possible, you ask? Well, what's hidden from view is the automatic "make-up gain" that is applied and is directly related to the threshold setting. If you lower the Threshold 2 dB, the output is increased 2 dB. This is helpful when you want the overall 'average' level of the mix to increase, but it is deceptive since you can't really properly hear how the mix is changing. The only way to hear the difference is if you compare it to the original in a level-matched way!



OZONE TIP

The "Automatically Match Effective Gain" feature has Ozone determine how much gain is being added by all of the active Ozone modules, and then automatically adds this amount of gain when Ozone is put into bypass mode. Basically, it does the hard work for you and you don't need to worry about precisely setting the "Gain when bypassed."

The appropriate range for the Threshold depends on the levels of your mix. For a subtle bit of limiting, bring the Threshold down just past the "crest" of the incoming signal histogram. This will limit the peaks above the green threshold line. In a mix with a reasonably strong level, try a Threshold that gives you 1-3 dB of limiting.

In general, a Margin setting of –0.3 to –0.6 will be appropriate as a final output level for your mix, depending on how much processing will be performed on the mix after Ozone.

Mode

The characteristics of each of Ozone's five selectable algorithms are explained below:

Soft: The Soft limiting algorithm uses the Margin (or final output level) as a guide but not as a fixed limit for the output level. It provides a natural "soft" limiting effect at the expense of allowing the level to exceed the margin.

Hard (formerly Brickwall): this uses the Margin point as an absolute guide, and the final output level will not exceed this point. Both the Hard and Soft algorithms provide a natural "analog" limiter effect, so the choice is related more to the behavior of the limiter and the output level than a choice in sound quality.

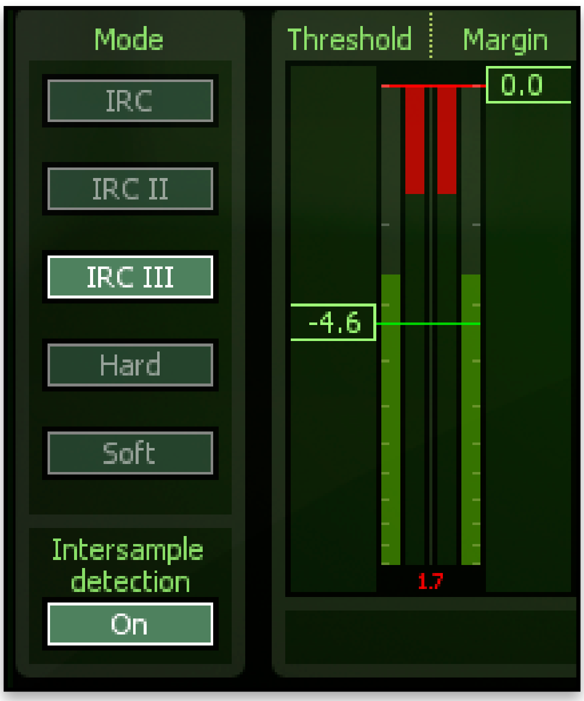
Intelligent (IRC I, II and III): This algorithm provides intelligent digital loudness maximization of the signal. Unlike the analog-modeled limiters (Soft or Brickwall), the Intelligent processor is designed for neutral, or transparent, limiting. It does this by analyzing the source material and applying limiting in a psychoacoustically pleasing manner, reacting quickly to transients (to prevent pumping) and reacting more slowly to steady bass tones (to prevent distortion). IRC I, II and III differ in the amount of protection from distortion they offer, but also in the amount of CPU usage, both of which increase from IRC I to II to III, respectively.

Character

The Intelligent (IRC) limiters provide intelligent release controls, so when using the Intelligent mode the release slider is replaced by a Character slider. This control allows you to modify the behavior or "character" of the limiter, indicated by the label above the slider. For optimum sound quality, the algorithm is constantly adjusting its response to the incoming material using the weighting of the Character slider. For example, if an extremely transient passage (such as a drum fill) suddenly hits the limiter, IRC will be able to actively minimize distortion.

Intersample Detection

In the digital world, audio is represented and processed as discrete individual samples or levels. When played back in the real world, these samples are converted to continuous waveforms. In some situations, the nature of this "real world" audio signal could result in clipping "between the samples," even when the limiter is limiting the digital samples. When working with IRC algorithms, selecting the "Intersample Detection" option allows Ozone to intelligently predict the behavior of the analog signal reproduced for the listener, and prevent any intersample clipping from occurring in the analog domain.



GENERAL MAXIMIZER TIPS

- Use the intelligent (IRC) modes for transparent limiting, the Hard mode for analog limiting with a fixed threshold and the Soft limiter for gentle boosts.
- Do not set the Margin above –0.3 dB. Technically, you can set the Margin to 0 dB so that the output of Ozone is maximized to the point of clipping but there is a good chance the sound will distort when played through a consumer playback system.
- More aggressive loudness maximizing (lower Threshold values) will generally require longer release times.
- ALWAYS listen to the music before and after the limiter in a level-matched way, just to be sure you are not causing unwanted artifacts. Remember, checking the "Automatically Match Effective Gain" box in the Ozone preferences will automatically volume match any A/B listening comparisons that you do.
- The digital look-ahead limiter is a very powerful tool. What this means to you is that you can do extreme loudness maximization and some of the distortion or artifacts will be initially hidden from the listener. Just consider whether that's what you want or not. Dark Side of the Moon is one the best-selling CDs of all time, and it used very little compression or limiting. More loudness means less dynamics (emotional highs and lows). There's always a tradeoff.

8: REVERB

WHAT'S THE GOAL OF REVERB WHEN MASTERING?

If the mix engineer has done a good job with imaging and reverb on the individual tracks and as a result you have a cohesive sense of space, you probably won't need to add any additional reverb to the final mix. In some cases, however, a little mastering reverb can add an overall depth to the sound. For example:

1. A recording made "live" in an acoustic space might have troublesome decays or room modes. In this case, a coat of reverb to the final mix can help smooth over any imperfections in the original acoustic space.

2. A short reverb can add fullness to the mix and improve bass intelligibility. In this case, you're not trying to add an impression of reverb to the mix, but instead creating a short reverb at a low level that fills in the sound stage.

3. In some cases, you don't have a good sense of ambience or cohesive space in the mix. Each track or instrument might have its own space, but they don't seem to gel together in a common space. Mastering reverb can be used as a way to unify the collection of recordings.

4. Sometimes the mixes have been cut too short, and the mastering engineer doesn't have a long enough audio tail to do the required fade out. A slight application of reverb at this point can help extend the natural decay of the last chord of the music, for example, allowing for a more comfortable fade out.

PRINCIPLES OF REVERB

In the simplest sense, a reverb simulates the reflections of sound off walls by creating dense echoes or delays of the original signal. In the same way that sound behaves in an acoustic space, the synthesized delays or reflections of a reverb decay over time. In addition, as the signal is delayed or reflected over time, the density of echoes increases over time (although decreasing in level) and you hear a "wash" of sound as opposed to individual echoes.

There are many types of reverbs, from plates to springs to reverse reverbs to gated reverbs. In the context of mastering, you can separate reverbs into two categories: Studio Effects and Acoustic Simulations. These aren't technical definitions, but more a way of thinking about reverb.

Acoustic reverbs simulate a realistic acoustic space. For placing individual performers (tracks) in a virtual room, these are excellent choices. You can clearly hear the "early reflections" from the original signal echoing off the nearest walls, and decaying into a space with later reflections. You also have a clear sense of the "positioning" of the track in the room.

Studio reverbs, on the other hand, are effects that aren't necessarily based on a real acoustic phenomenon. While they might not sound as natural as an acoustic reverb, they have been used so much on commercial recordings that we have come to accept (and even expect) them. Do they sound like a real room? No. They are an effect of their own, and they can give an overall sheen or "lush" ambience to a song. You don't picture the musicians performing in a real acoustic space, but instead experience a wash of ambience. Use caution: it is easy to overdo it and it can wash your mix right down the drain, but just a touch can wash away any imperfections in the original mix and give it a nice sheen.

USING THE OZONE REVERB

Ozone provides both acoustic space and plate "studio-style" reverbs that you can apply to your mixes. The reverbs use hybrid processing, utilizing both convolution and advanced algorithmic technology.

Ozone's reverb does not include any gate, reverse or other "special effect" reverb controls. Those might be great for individual tracks, but they are not for overall mixes. Think of Ozone's options almost as a "coating" or "final polish" reverb that lives atop any per-track reverb.

The best way to become familiar with the sound is to load up a song, solo the Reverb module (so you only hear the effect of the reverb processing) and solo the Reverb's wet signal so you don't even hear the original direct mix. You only hear the Reverb. To do this, click "Solo Wet," as shown below:

Turn up the Wet Mix fader. This controls the amount of reverb that is being mixed back into your mix.

Adjust it to a comfortable listening level to go through this section of tutorial, which will inevitably be much higher than what you would want if you were actually adding reverb into your master.

Mode

These buttons allow you to select between the acoustic room reverbs (Room and Hall in Ozone 5 with Ozone 5 Advanced adding Theater, Cathedral, and Arena) and the studio Plate reverb. As mentioned above, the plate mode provides a lush, smooth sound while the room modes provide a more natural, acoustic sense of space, be it small or large.

Pre-Delay

Pre-Delay sets the amount of delay in milliseconds between the original signal and the beginning of the reverb. Consider the acoustics of a large room—you'll only hear the first bit of reverb after the original signal has gone out to a wall, bounced off it, and come back to your ears. The length of time before you hear this initial reflection is controlled with the Pre-Delay slider.

Decay Time

In an "acoustic" sense, this controls the overall size of the room—or, to be technical, it controls the "decay time" of the reverb. Higher values will give longer reverb times, as it will take longer for the sound to decay.

- If you're trying to "wash over" a mix, you'll probably want to try values in the range of 0.80 to 1.20 for this fader. As a general tip, if your mix already has reverb on the individual tracks (which it probably does), try to set the Decay Time or length of the reverb to slightly longer than the reverb on the original tracks. You can always adjust the level of the mastering reverb with the Wet slider, and a longer decay time on the overall mix will blend things together better. In general, if we're going to apply mastering reverb, we usually end up with Wet around 5.0 to 15.0 (and Dry at 100.0)
- Another interesting effect to play around with is to use a small room size, anywhere from 0.25 to 0.60, and turn up the Wet slider a little more to 20.0 or 30.0. In some cases this can create a fuller sound by adding a short reverb, or doubling, to the mix. It can also make some mixes sound terrible. (Listen before you send it to the duplicator!) You'll also want to keep the reverb Width at 100.0 if you use this effect, since spreading out an extremely short reverb wouldn't be very natural. In effect, you'd be creating a small room with wide walls, which just doesn't make sense (or sound good).

Frequency-Specific Decay Controls

In a real room, the sound decays as it bounces off the walls. Decay Time sets the amount of time it takes for the reverb to fully decay. However, not all frequencies decay at the same rate. A padded cell would decay the high frequencies faster than a bathroom. Different rooms and wall materials have different absorption properties, and the High Decay control lets you control the characteristics of the high frequency decay of the signal. Likewise, the Low Decay setting adjusts how quickly the low frequencies decay.

- Greater High Decay values will result in a brighter-sounding reverb. Lower values are, well, less bright.
- Many people typically use Ozone with the Low Decay and High Decay settings around 1.00, but experiment with these sliders if you're looking for unique reverb effects.

Width

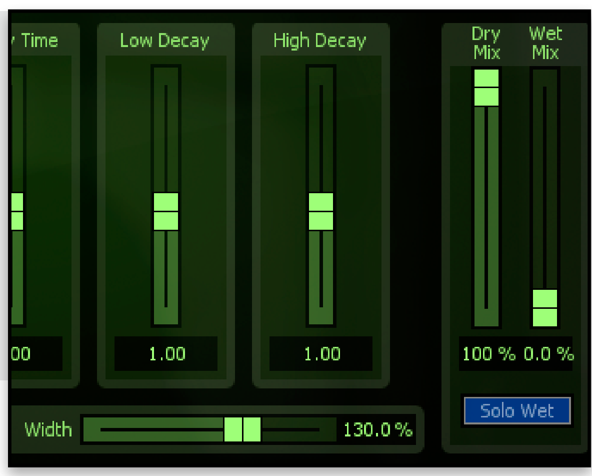
The Ozone mastering reverb doesn't return the same reverb signal in the left and right channels, as this would sound unnatural, and doesn't approximate what would happen in a room.

Instead, it creates a nice spacious "diffuse" sound by returning slightly different left and right channels of reverb. The Width slider lets you control how different the left and right channels will be.

In an "acoustic" sense, you perceive this as the width of the room, or at least the width of the reverb signal.

MASTERING QUICK TIPS

- In most cases, you'll want the width to be from 100 to 150.
- As you turn up the Width, you'll tend to perceive more reverb. At higher room widths, try turning down the Decay Time. This might seem counterintuitive, but give it a listen (turn up the width to an extreme of 200) and you'll hear what we mean. The ideal balance is, well, a balance between the two.

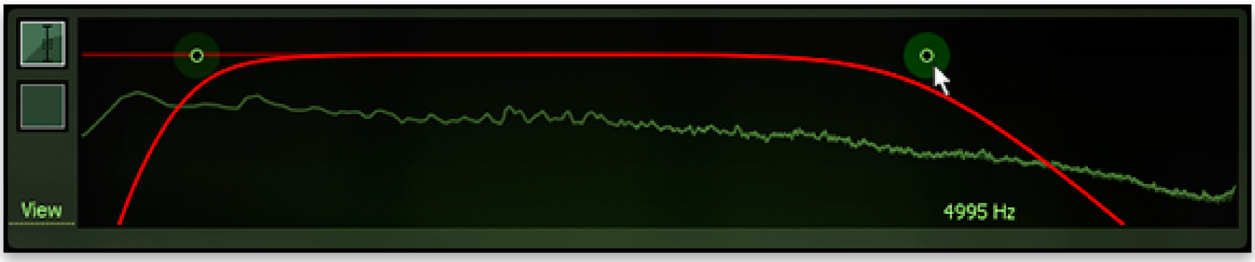


High and Low Cutoffs

You may have noticed that the mastering reverb has a spectrum with two node controls. Placed on the outer edges of the spectrum by default, these controls do not function in the same way the node controls in the EQ do. Instead, in this module, they roll off of the reverb signal at different frequencies.

You can drag the nodes to the left or right to change the bandwidth of the reverberated signal that is returned and mixed back into your mix. The area under the bright red line will be the reverb signal that you hear.

So where should you put the cutoffs? Well, the mastering reverb in Ozone rolls off with high frequencies by design, so you don't necessarily need to roll off the high frequencies yourself. At the same time, rolling off the highs (moving the right node to the left) can take away some of the "tinny-ness" of the reflections, and rolling off the lows (moving the left node to the right) can take away some of the rumble of the reflections.



We tend to start with the low cutoff at 80 Hz and the high cutoff around 3-5 kHz. If we hear "sibilance" (too much "ssss" and "shhh") from the singer, we move the high cutoff down below 2 kHz, as high frequency reverb can accentuate sibilance in an undesirable way.

GENERAL REVERB TIPS

Like any effect, it's easy to overdo reverb. Unless you are looking to add reverb as an effect on a recording, using a "wet" amount of 6-8% is the maximum you would want to use, and something more in the region of 2-3% is typically more appropriate.

Here are some tips for "keeping it real":

- Bypass the mastering reverb from time to time to get a reality check on what the dry world sounds like. In most cases, a reverb should be "sensed" more than it's heard, if it's used at all on a mix.
- If you want MORE reverb, keep in mind that you have multiple options. You can increase the wet amount (the level of the reverb mixed into your mix), or you can increase the Decay Time (the size of the room) or you can increase the room Width. Adjust each of these, then use the History window (or A/B/C/D feature) to decide which adjustment was the most effective.
- Though you can reorder where the reverb is applied in the signal chain, my experience is that it works best last. EQ and dynamics processing become much more obvious if they are applied to a "wettened" signal.
- Compare to commercial mixes for a reality check. What to compare against depends on the sound you're shooting for. Something like Steely Dan is pretty dry (and more of an acoustic room reverb) where something like George Michael or Phil Collins can be very lush (with more of a studio plate reverb sound). If you've got a pop ballad, you're probably going to be able to get away with a thicker coat of reverb than on a hip-hop mix.
- If you're applying a wide reverb (room width up between 100 to 200) keep an eye on the phase meters, and use the Channel Options (especially the Mono switch) to check to make sure it doesn't completely fall apart in mono.

9: STEREO IMAGING

The "image," or "spatialization" of your mix is like a sonic picture, comprised of the overall panning and spatial placement of individual components within the stereo mix. In the context of mastering, "stereo imaging" refers to the manipulation of the perceived image of a mix to enhance the listening experience. Sometimes this means "widening" a mix, but it equally can mean "narrowing" a mix to solve certain problems, as we'll discuss shortly.

WHAT'S THE GOAL OF STEREO IMAGING WHEN MASTERING?

Stereo imaging is a tough task—it's typically only used in specific situations and often very gently. It's difficult to get a cohesive mix that still has a sense of space and imaging. Usually the over-application of effects makes it all the more difficult to "image." Manipulation in the stereo field is a little like salt—a small amount might taste good on food, but more than a little and that's all you taste. Perhaps more than any of the standard mastering tools, this is the easiest to overdo. However, stereo imaging can also be useful as a corrective tool. For example, it can help re-center low and bass-y frequencies that might otherwise cause phase and mono compatibility problems, particularly when audio is played back via mono sound sources (such as live venues).

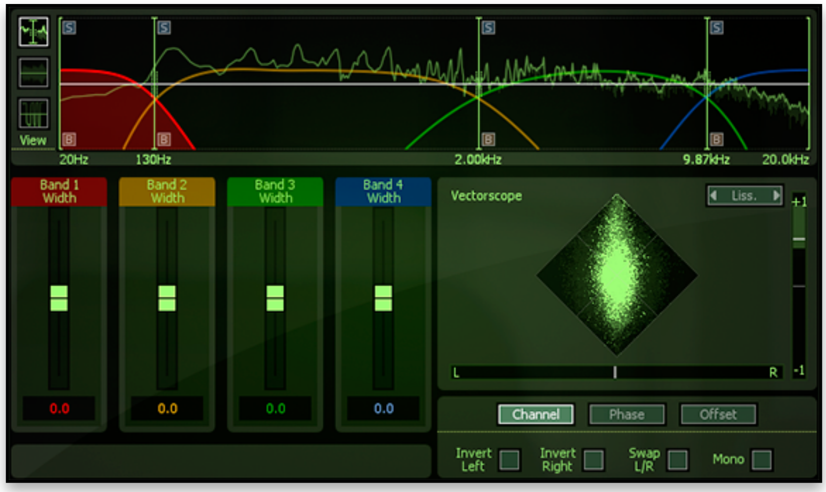
PRINCIPLES OF STEREO IMAGING

Most pop/rock-based musical idioms have the following in common: the most important elements are the drums and the vocals. To that end, the kick, snare, and lead vocal tracks are usually panned to the center. When you use a stereo widener, you are therefore usually emphasizing the "other" elements in the mix. Yes, a little of that might help, but I mean a little.

Other issues come into play regarding phase relationships and sonic clarity overall, so if you use a widening tool, listen to be sure that the "heart" of your recording isn't diminished.

USING THE OZONE STEREO IMAGER AND PHASE METER

The phase meter indicates the degree of similarity, or "correlation," between the left and right channels.

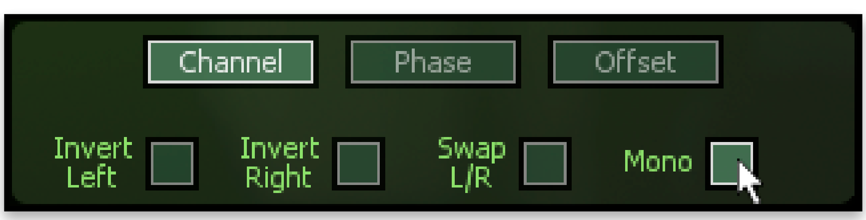


When the audio in the left and right channels is similar, the meter draws towards the top. The extreme case is when the left and right channels are exactly the same, in which case the correlation is +1 and the meter would be positioned all the way to the top.

When the left and right channels are uncorrelated, or very different, the meter draws towards the bottom. The extreme case here would be for the left and right to be exactly out of phase, in which case the correlation is -1 and the meter would be positioned all the way to the bottom. As the phase meter updates, it "paints" a history to show the correlation of the left and right channels over time. Brighter regions indicate that the phase meter has spent more time in that area. This provides you with a quick way to visualize the extremes of the phase correlation, as well as the most common regions. Note that you can reset the region drawn by the "phase needle" by clicking on the meter.

In general, most recordings have phase correlations in the 0 to +1 region. A brief readout towards the bottom side of the meter is not necessarily a problem, but could represent a possible mono compatibility issue.

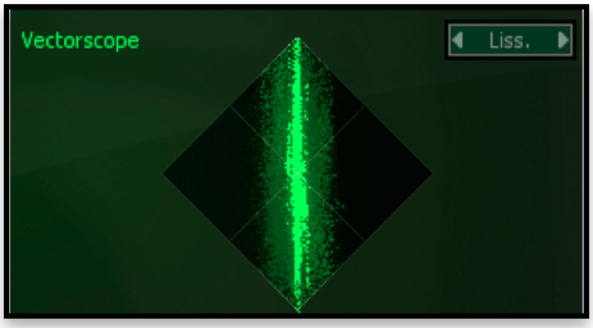
You can perform a quick check of mono and phase compatibility by clicking on the "Mono" box in the channel settings, which will sum the left and right channels to mono. Other menu options are to invert the polarity of left or right channels, and swap left and right channels.



As you apply greater multiband stereo widening to your audio, the phase correlation will tend to draw more towards the bottom, as the left and right channels will become "wider" or less similar.

By default, the phase meters are placed at the end of the signal chain so you are "seeing what you hear." A useful side effect of this is that as you mute bands, the phase meter displays the stereo correlation only for the band(s) that you're hearing. You therefore have a multiband phase meter that lets you analyze the imaging for individual bands.

Vectorscope



The Vectorscope also provides a view of the stereo image of the signal. Typically, stereo recordings should be a random pattern that is usually taller than it is wide (as shown in the screenshot). Vertical patterns mean that left and right channels are similar (approaching mono, which is a vertical line). Horizontal patterns mean the two channels are very different, which will sound wider but could result in mono compatibility problems.

Below are some Vectorscope display options:

- You can click on the phase meter to reset the peak hold display.
- If you want to turn off the peak hold display, you can turn it off in the Options Screen.
- Multiple operation modes of the Vectorscope:

    Lissajous: The Lissajous Vectorscope ("Liss.") plots per-sample dots on a traditional oscilloscope display. Typically, stereo recordings produce a random pattern on a Lissajous Vectorscope that is taller than it is wide. Vertical patterns mean left and right channels are similar (approaching mono, which is a vertical line). Horizontal patterns mean the two channels are very different, which could result in mono compatibility problems.

    Polar Sample: Like the Lissajous Vectorscope, the Polar Sample Vectorscope plots dots per-sample, but uses a polar coordinate display that is more useful in highlighting the stereo image of the incoming signal. Patterns that appear within the 45-degree "safe lines" represent phase coherent signals, while patterns outside these lines represent out of phase audio.

    Polar Level: The stereo energy of a recording is clearly represented by the Polar Level Vectorscope, which plots rays on a polar coordinate display that represent sample averages. The length of the rays represents amplitude while the angle of the rays represents their position in the stereo image. Rays within the 45-degree safe lines represent in-phase audio, while anything beyond these lines represents audio that is out of phase.

GENERAL STEREO IMAGING TIPS

- You can generally do more widening of higher bands, as opposed to lower bands, without destroying the clarity of a mix.
- You may even want to try narrowing of lower bands to pull bass to the center.
- Monitoring on headphones is going to give you a false impression of the imaging of your mix. You really need to check imaging on speakers. Headphones will always feel "wider" because none of the right channel is being heard in the left ear or vice versa.
- Keep checking mono compatibility with the Channel Ops menu, or via a mono summing device.
- Ozone supports automation, which allows you to change values of controls over the course of a mix. One trick is to automate the widening of a mix—i.e. widening the mix on a chorus then tightening it back up for the verse.

OZONE TIP

If you happen to be using the Advanced version of Ozone, the "Stereoize" feature in the Stereo Imaging module offers the ability to widen even entirely mono signals! This can be used to great effect, particularly for widening and giving new life to vintage, mono recordings.

10: HARMONIC EXCITER

An exciter can add "energy" to an audio signal and, in the context of the mastering domain, to an entire mix. This doesn't necessarily mean increasing the volume, or boosting the signal, however. A sense of "energy" can be created by adding additional combinations of odd or even harmonics to the frequency content of a mix, and that's what a harmonic exciter is designed to do.

WHAT'S THE GOAL OF HARMONIC EXCITING WHEN MASTERING?

If one of your mastering goals is to add power, punch, and other subjective terms such as "warmth" and "brightness," the Harmonic Exciter might be a good tool to use.

There are many design strategies behind exciters, from dynamic equalization, waveshaping and distortion to short multi-band delays. When people speak fondly of recording music to tape machines, or reminisce about the sonic characteristics of tube equipment, part of that enthusiasm is a result of the harmonic distortions that these and other types of analog equipment impart on audio, as further described below.

How could distortions (a sometimes-dirty word in audio) be so appealing? Well, in truth, distortion in small doses isn't necessarily a bad thing. If designed correctly and applied with restraint, distortion can introduce harmonics that add excitement or sparkle to a mix.

As an example: Ozone provides a selection of exciters modeled on tubes, triodes and tape saturation. When tubes saturate, they exhibit a type of harmonic distortion that is generally described as "warm" or "musical." If used with care, this distortion creates additional harmonics that add presence or sparkle to the mix while still preserving a natural characteristic. You can perhaps see why boosting high frequencies with an EQ is not going to achieve the same effect. Boosting an EQ simply turns up the existing harmonics, whereas a Harmonic Exciter can actually synthesize *additional* harmonics. Tape saturation provides a similar effect, although the harmonics that are created are more "odd" than "even." That is, tube saturation typically generates even harmonics that are an octave apart (again, "musical"), while tape saturation is a slightly more aggressive excitation, generating odd harmonics that are a fifth apart.

PRINCIPLES OF HARMONIC EXCITATION

It's very easy to overdo an exciter. What may sound good at 3.0…might sound even a little better at 4.0…and once you get used to that, you find yourself pushing it up to 5.0 to keep the "excitement." Before you get caught up in the excitement (pun intended, we guess) and send it off to the duplicator or to SoundCloud, do a little reality check:

1. Compare it to some commercial mixes. Okay, in some cases these are overdone as well, but it depends on the genre and sound you're shooting for. What works for a dance mix probably isn't going to sound as appropriate on an acoustic jazz number.

2. Live with the "excited" mix for a while. At first listen an exciter is, well, exciting, but over time it can really sound fatiguing or even harsh and annoying.

USING THE OZONE HARMONIC EXCITER

This is a very easy effect to use. (That could also be why it's often overused.)

As a multi-band effect, the Exciter has up to 4 bands to work with, and each has a pair of controls. In most cases, you're going to use the Amount control. In addition, you'll probably apply excitation to the upper one or two bands, although there are some cases where tube saturation in small amounts across the entire spectrum can be musically pleasing.

With your mix playing back (of course), adjust Band 3's Amount slider upward. As you move the slider up, you'll hear what starts as sparkle and excitement, but it can quickly turn against you as you go up too far. Take note of the point where it starts sounding "annoying" and then turn it back down to 0.0. Note that higher bands can usually bear higher amounts of excitation.

OZONE TIP

Clicking on the 'Oversampling' button engages more processing power. This will greatly increase the accuracy and sound quality of the analog modeling in the harmonic exciter. If your computer is powerful enough, you may run this in real time. Otherwise, it's good to engage this tool before rendering/bouncing.

OZONE TIP

As you work with these effects, you can use the checkbox labeled "B" to bypass any multi-band processing applied to that band by Ozone. As a counterpart to the Solo checkbox, the Bypass checkbox is a useful tool for hearing what sections of your mix are being processed through each band. Note that this Bypass applies to ALL multi-band processing, including multi-band harmonic excitation, stereo imaging, and dynamics.

OZONE TIP

You can right-click on the multi-band spectrum display to remove or insert bands. For additional guidance, there is also a right-click and "Learn" feature that will automatically analyze your audio signal and determine the most appropriate, transparent places to place your crossover points.

GENERAL HARMONIC EXCITER TIPS

- In most cases, excitation to the upper bands will give the desired effect. However, since Ozone offers an analog tube saturation model for the harmonic excitation, you can achieve a type of "tube emulation" on the low-mid bands as well. For this scenario, you'll want to try a very small amount of equal excitation across all bands. (In other words, keep the Amount control low and work in single band mode.)

- You can get a "dirty" bass effect by applying some excitation to the low band. If you're simply going for more bass level, use an EQ or maybe the Dynamics module, but if you're looking to add grunge on those bass harmonics, sometimes you can turn to the Harmonic Exciter. You can also check out iZotope's Trash for extreme multi-band distortion fun (**www.izotope.com/trash**).

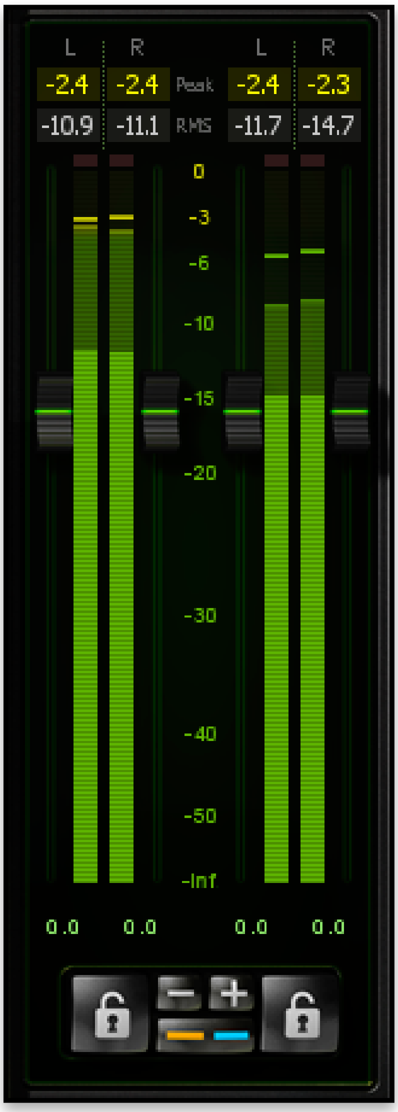
- The Tape excitation mode provides a bright, saturated tone; the Tube model provides a thicker midrange rich tone; and the Retro model provides a heavy saturation character. The Warm exciter mode is similar to Tube, but is unique because it generates only even harmonics that decay quickly, and you can use Warm to drive tube style saturation quite hard, without creating "mud" in the low mid-range.

- Electronica mixes, or mixes that are comprised of more soft synths and samples than acoustic instruments, can sometimes sound a little lifeless. Use the Harmonic Exciter as described above to change this!

11: METERS

Here's a list and description of various sorts of meters and their uses in mastering. They are included within the Ozone suite and also in standalone products such as Insight.

LEVEL METERS



Level meters are probably the most familiar and ubiquitous meter. They will usually display Peak level (the level of a signal from moment to moment) and RMS or "average" level (the level of a signal averaged over a short window of time). Both types of information are important but for different reasons.

Peak level tells us how close the signal is to the point of distortion. It's a way of helping us understand whether we have any headroom to bring a signal up without changing anything else about it and staying below the point of distortion.

RMS level gives us information that relates to our perception of volume. Our brain processes information during a short window of time to evaluate how loud something is in our environment, and an RMS level is a way to attempt to give feedback about that. However, RMS doesn't relate directly to perception in the sense that it doesn't take frequency content and balance into account.

The relationship between Peak level and RMS level will vary widely depending on the dynamics of the mix and the genre of the music. That makes it hard to generalize too much. However if one had to give a general idea about where the RMS level should generally sit with respect to 0dBFS:

Electronica: -8 to -12

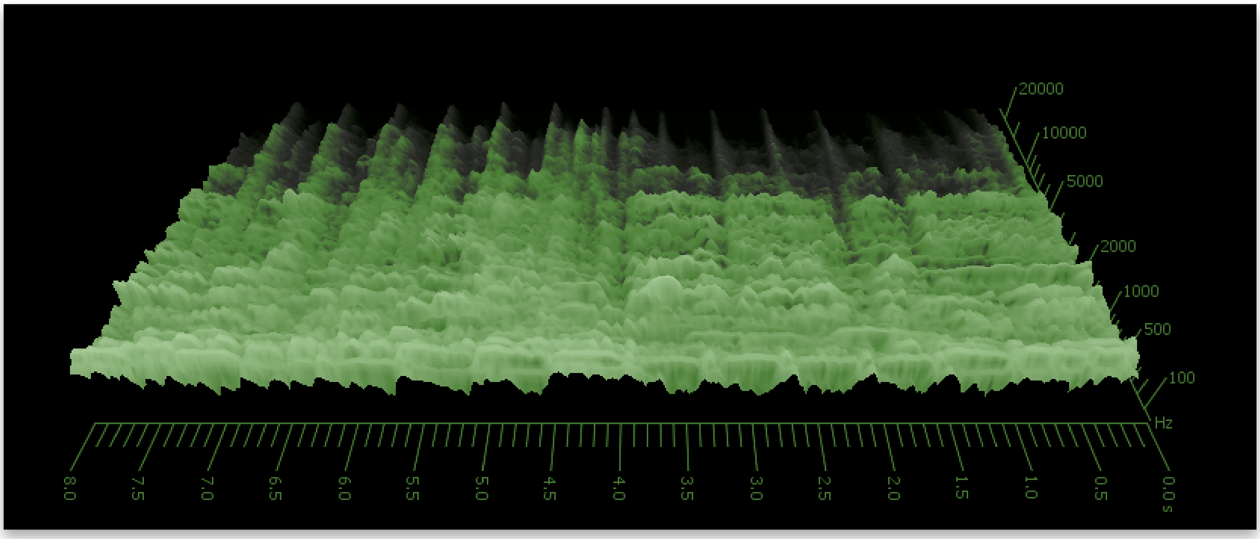
Pop/RnB: -10 to -14

Rock: -12 to -16

Acoustic idioms (Jazz, Classical, folk music related): -14 to -20

And that brings us to….

SPECTROGRAMS



While not exactly a meter, a spectrogram does map levels, or energy, across the spectrum. It is a helpful tool that provides a reality check on what you hear **and**, to an extent, what you *can't* hear—especially in the nether regions of the bass and the region close to 20kHz. Spectrograms are good for helping you quickly diagnose a problem frequency or set of frequencies. For instance: when a track that has a problem with sibilance (the sound a vocal 'S' makes), it usually shows up quite readily on a spectrogram. That makes it easy to focus an EQ or a de-esser on the problem.

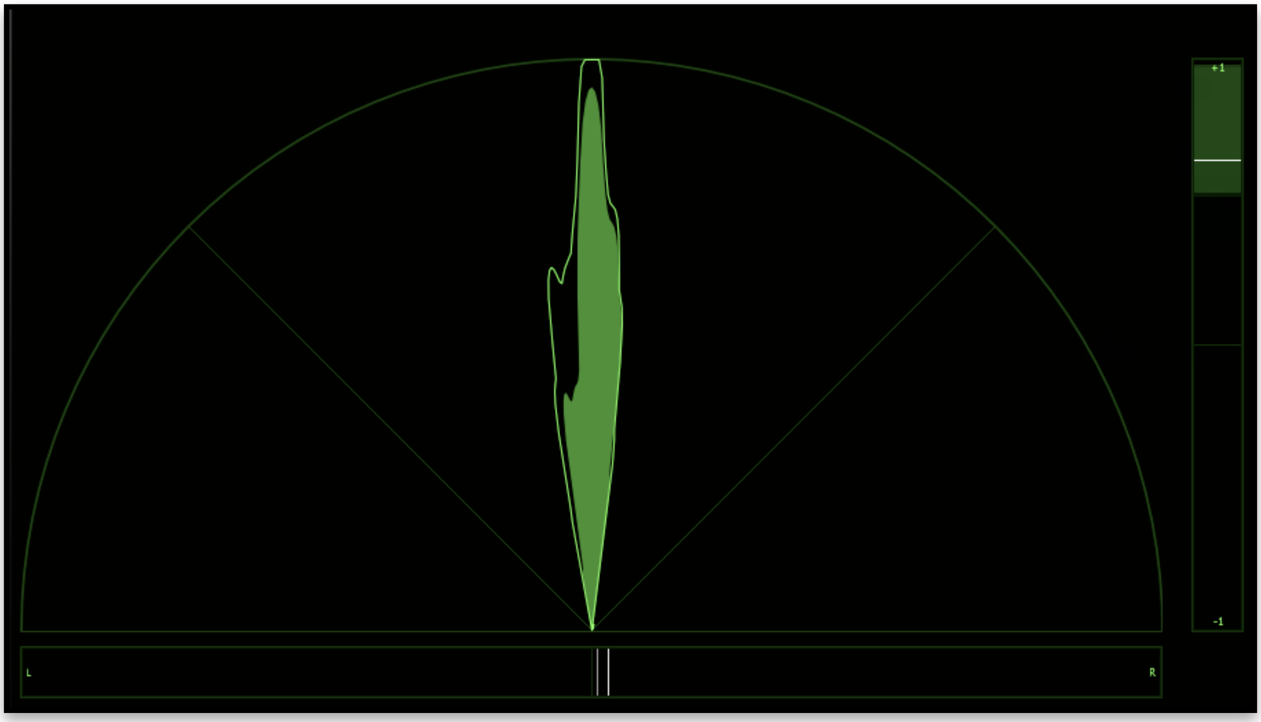
It's helpful to have a spectrogram running while working in the EQ module to help you focus your EQ settings. It's also useful to have one running at the beginning and the end of your mastering chain to keep tabs on what is happening and how your original mix has been altered.

VECTORSCOPE/CORRELATION/STEREO IMAGE METERING

This sort of metering is probably the most under-appreciated, and in some ways this isn't surprising. The concept of thinking about mono-compatibility, and the width of a stereo image, isn't always discussed often enough when people begin to learn about audio engineering or mastering. But we won't let it pass by here!

When you are mastering, you want to be sure that the main instruments in a mix don't disappear when you listen to it in mono. A correlation meter gives you visual feedback so you can be sure that the recording has a strong orientation toward mono, and that it most certainly does not spend too much time with a strong orientation toward out of phase information. Why? Scenarios such as vinyl cutting, MP3 encoding and terrestrial broadcasting (radio and TV) rely heavily on the mono signal being in good proportion. In any of these scenarios, too much out of phase information will cause unpleasant artifacts for listening and, in some cases, actually cause a mix to be sent back to the mixer as unusable.

As with any mastering process, a major problem with phase is best corrected at the mix stage. If that's not possible, one can try to address these problems using mid-side processing or the stereo imaging tool. In these cases, the correlation meter is a helpful gauge for making your adjustments.



12: A NOTE ON TARGET FORMATS

A common question for any budding mastering engineer is about format. Do I master for CD the same way as for MP3 or vinyl or whatever the next big thing is going to be?

A mastering engineer needs to be well-versed in musical styles and the tonal and dynamic aspects of each style. In a similar way, the mastering engineer needs to understand the different methods of delivering end product to the consumer. When this is understood, the mastering engineer can prepare the audio as skillfully as possible for the next step.

Vinyl cutting, MP3/AAC encoding, CD replication and broadcast each have their own set of technical issues, and each have implications for how one processes and delivers the final audio. There isn't room to dive into each in depth, but here are a few tips to consider:

Vinyl: when preparing audio to send to a cutting engineer, the transient peaks should be maintained. As a result, use little or no limiting.

MP3/AAC: These encoders invariably add to the signal level due to the intense processing required in the creation of the audio file. That means the peak level of the source WAV/AIFF file might need to be lowered by as much as 1 to 1.5 dB before encoding.

Broadcast: Broadcast chains usually include significant dynamic range processing. Avoid hyper compressing or limiting since there's another compression stage coming. Some people think they need to compress heavily in order for their recordings to sound good on the radio. The converse is true. Make it sound good, with good dynamics, and it will survive broadcast processing better.

13: TIPS FROM MORE PROS

We asked several seasoned veterans for a few words about mastering. Some offered specific comments on specific tools. Others gave a sense of how they approach the music itself. In each case there are valuable insights to be gleaned.

GREG CALBI

www.sterling-sound.com

Credits Include

Bon Iver, Alabama Shakes, Norah Jones, Grouplove, MGMT

> *First thing, if the client is present, is ask which he/she thinks are the best mixes from his/her project. This tells me whether or not I'm hearing things the same way he/she is. I learn a lot from this simple question.*
>
> *Then I ask him to tell me which ones were the most difficult. This will usually reveal if the client is happy with what he has overall. It tells me whether or not I start with a conservative approach to the project or a more aggressive one.*
>
> *If the client is not present, I usually ask for a simple e-mail with any information the client thinks I should know going into the mastering. If there are no comments from him, I can assume that he is fairly content with the mixes he has. If the notes are extensive, I try to analyze what type of listener I am dealing with, and how technical he is. This will help in my dialogue with him post-mastering. If he just says things like "make it rock" or "kill it," I take that as code for participation in the now-famous "level wars." In either case, I will then listen to a minute or two of each mix to determine if there is a common thread, or if there is a great variety of production ideas, different mixers, studios, instruments, etc.*
>
> *This sets me on a course for my approach, and for determining how long the mastering will take. I also take careful note of the mix levels, and if some of them are already extremely compressed and over-saturated, I need to approach the album with those same levels as a goal (plus just a little bit more).*

ADAM AYAN

Mastering Engineer at Gateway Mastering in Portland, ME

www.gatewaymastering.com and www.adamayan.com

Credits include:

Carrie Underwood, The Dandy Warhols, Foo Fighters, Madonna, Sarah McLachlan

> *I suggest starting any mastering session by focusing on the tonal shaping and equalization of the recording. I feel that mastering has, in the mind of many, become so much about compression and getting a mix loud—often at the sacrifice of proper tonal balance. Spend as much time as needed getting the EQ curve just right first, then dig into compression/limiting/loudness. With some experience you'll find that these things (EQ and loudness) do naturally go hand in hand, and affect each other greatly. But until you master proper EQ skills, I suggest focusing on the EQ first.*

BOB OHLSSON

Owner of Bob Olhsson Audio Mastery in Nashville, TN

www.audiomastery.com

Credits include:

Marvin Gaye, Stevie Wonder, Keb Mo, Michael Stearns, Robert Rich

> *I think one of the biggest challenges in recording and mastering is finding the signal processing that the recording requires, as opposed to what only enhances how something sounds in one room with one set of speakers. The more complex the processing, I've always found, the more likely it is that my results may not travel. I think mastering is often a matter of trying to correct these kinds of problems that were created in recording and mixing. I'm often looking for that single move in EQ that pops the mix right into focus, where the mixer believed he/she had it in the first place.*
>
> *I'm also concerned with creating a master that isn't fragile and won't fall apart when it hits a multiband broadcast processor. Putting two multibands in a row often tells the tale of why you don't usually want a mix passing through one prior to that final time at the radio or TV station.*

One has to be very careful with any sort of Mid-Side stereo "enhancement" because in the real world, FM stereo is usually really somewhere between stereo and mono depending on how far you are from the transmitter. Likewise plenty of venue systems are still mono, as is low bit-rate streaming. The idea is for the mix to sound great everywhere and not just in stereo. Stereo "enhancement" can sometimes help, but must be used with great care to make certain it isn't degrading the mono sound.

MARC-DIETER EINSTMANN

Founder of Einstmann Mastering in Hamburg, Germany

Credits include:

Mary J. Blige, Notorious B.I.G., Depeche Mode, Yo-Yo Ma, Elvis Costello

A few observations on using exciters: In a mix, exciters are most effective when applied to individual instruments. In essence, an exciter adds gain to the harmonics of an audio source. Exciters may be modeled after the distortions we know from "vintage analog" circuits, such as different types of output transformers, tubes and magnetic tape. Other architectures may be completely new and "exciting." The danger here is to color your audio source unnaturally and saturate the audio to where there is little room for correction at the mastering stage. The effect becomes even worse when these exciters are applied across the stereo bus, as these are processes that cannot be undone without penalty.

At the mastering stage, if absolutely necessary, exciters should be used in multiband rather than broadband mode to minimize potential adverse effects of the process. There are only a few exceptions. If in doubt, leave exciters out of your stereo bus.

SCOTT HULL

Owner & Mastering Engineer at Masterdisk Studios in NYC

www.masterdisk.com

Credits include:

Sting, John Mayer, Herbie Hancock, Donald Fagan, Steely Dan

Mid/Side processing can be very finicky…you have to listen very carefully in an excellent stereo environment to be sure you are not doing more harm than good.

There are often "hidden" side effects to processing M/S. One important tool you should have is the ability to monitor the Mid signal separately from the Side. (Editor's note: fortunately this is something you can do within the Ozone plugin modules!)

Note there are not two "sides", only one. And for most engineers that don't have a minor in AC electronics…it's "magic" how the left and right channels re-appear when converted back to Left and Right. This process of flipping the phase and summing to mono is so important, it's built into almost any serious mastering monitor section. You can do this at the monitor for a quick listen. Again—you simply reverse the phase of the left channel then sum to mono.

You will notice that there is usually a lot more energy in the Mid signal than the Side. If you are processing Gain via a compressor, you may not have enough gain or threshold range to properly compress both signals…but that is ok—it's often nice to leave the Side alone – this prevents too much pumping.

What I find I use the M/S EQs and level controls for:

- *Bringing the track up—in general to complement a "too loud" vocal.*
- *Slightly brightening the Side signal to enhance the stereo width perception.*
- *Raising the Mid if the Kick and Vocal need support.*
- *Taking mud out of a live mix by rolling off some low mids in the Side. Be careful, if you take too much, it starts to sound less "live."*

This all takes practice and careful listening. I don't like to say "Leave this to the pros…" but my experience says that if you don't use this tool with care, you will do more harm than good.

14: IN SUMMARY

We hope that this guide increased your knowledge of the art of mastering and, as a bonus, gave you some ideas on how to use Ozone effectively as a mastering tool. The difficult part of mastering (and trying to create a guide like this) is that every effect, setting, and parameter is entirely dependent on the content of the mix, the genre, the desired result, etc. With this in mind, we don't believe in products that fool you into thinking you can just select the "Hot Pop Master!" preset and you're done. If you know how to use Ozone, or your tool of choice, you'll be able to come up with a great sound on your own. And it will be your own great sound.

In the end, there are no right answers, no wrong answers, and no rules—just experiment and have fun!

Thanks,

iZotope, Inc.

P.S. Besides Ozone, we also invite you to try out our other products. Each is available for download as a free 10-day trial at www.izotope.com/trials.

Here's a quick overview of several that particularly complement Ozone:

IZOTOPE ALLOY™ | ESSENTIAL MIXING TOOLS

Alloy gives you futuristic tools, fast results and most importantly, fantastic sound. Bring character and life to every element of your mix with Alloy.

http://www.izotope.com/alloy/

IZOTOPE NECTAR™ | COMPLETE VOCAL SUITE

Make your vocals and dialogue sound professional in a broad range of genres with Nectar's complete set of vocal production effects.

http://www.izotope.com/nectar

IZOTOPE RX™ AND RX™ ADVANCED | COMPLETE AUDIO REPAIR

Fix previously untreatable audio with innovative visual editing and a complete set of repair tools. Remove noise, hiss, and clicks, resynthesize missing audio, and so much more.

http://www.izotope.com/rx/

IZOTOPE INSIGHT | ESSENTIAL METERING SUITE

An extensive set of audio analysis and metering tools, perfect for visualizing changes made during mixing and mastering, troubleshooting problematic mixes, and ensuring compliance with broadcast loudness standards.

http://www.izotope.com/insight/

APPENDIX A: GETTING SET UP FOR MASTERING

SOFTWARE AND SOUND CARD

To master on a computer, you will need some type of editing software and an audio interface or sound card. There are plenty of reviews and articles on software, sound cards, and audio interfaces, etc. that can help you find the equipment that will work best for you. One word of warning: you need to exercise caution when relying on the audio playback that is built into a computer, whether desktop or laptop. They are usually not designed for high fidelity and in some case will run the audio you hear through hidden signal processing. This will alter what you hear making it difficult to make good decisions during mastering. This is why we recommend acquiring additional equipment.

One important point is that when mastering, you're really just focused on improving a mixed-down stereo file. Some applications are optimized specifically for working with stereo files. However, you can also bring a stereo file into a multi-track program (i.e. Pro Tools, Wavelab, Sound Forge, and Adobe Audition, SONAR, SAW, Samplitude, Vegas, Cubase, Nuendo, Logic, etc.) as a single stereo track and process it that way.

Some of these DAW's incorporate the ability to ouput masters, with features for adding metadata and producing final product that can go straight to duplication, replication and distribution (Sequoia, Pyramix, SoundBlade, WaveLab, SADiE, DSP Quattro, Studio One 2, DDP Creator and others). Others do not. Depending on how far you want to take your mastered creation, you may need to investigate which software DAW or combination of software DAWs is right for you when you approach the task of mastering.

MONITORS

It's important that you monitor on decent equipment when mastering. If your playback system is giving you an inaccurate or incomplete playback of the sound, you can't possibly know what's the sound of the mix and what is coloration that's caused by your playback system. That doesn't mean you can't get decent results with relatively inexpensive equipment. The key is knowing the limitations of what you're monitoring on and learning to adjust for it in your listening.

For near field studio monitors, the most common problem is lack of bass, specifically below 40-50 Hz or so. These monitors just don't have the size or mass to move that much air at that low a frequency. One solution is to complement a pair of studio monitors with a subwoofer to cover the low end. However, it takes time and care to set up a sub-woofer so that it fixes more problems than it causes.

OZONE TIP

How do you do this? If you have a mic that's flat down to 20 Hz, here's a quick and dirty way to do it with Ozone.

1. Take a song with a good range of frequencies in it. We just happened to choose Vasoline (Stone Temple Pilots). As long as there's a broad spectrum, it doesn't matter which song (we did say this was the quick and dirty method).

2. Put Ozone's spectrum in average mode and loop a section of the song. Open the EQ module and save it as a snapshot (open the Snapshots tab, click "Start Capture" and you'll see a frozen line that's a different color than the active spectrum).

3. Place the mic in the spot where you would be listening from, and play the loop through the monitor/subwoofer combination. We used Cakewalk SONAR with "Effects on Input" enabled, so that we could see the result in real time.

4. Adjust the subwoofer level until the sound picked up by the microphone (the yellow line) is close to the spectrum of the source (the purple snapshot).

It's not exact and there are several variables here (the response and location of the microphone being the most significant) but it can get you close.

In the end, your monitoring environment is made up of more than your monitors. Probably the most influential part of the system (aside from your ear-brain connection) is the room. You'll never get a perfect listening environment, and you can never predict how what you're listening to will translate to all the systems out there that other people will use to playback your song. With that in mind, here are some tips we've picked up over the years for learning to master on studio monitors:

1. Listen to music that you know well and have listened to on many systems. Spend some time "getting to know" your monitors. Play your favorite CDs through them. You probably know how these CDs sound on a home system, a car radio, etc. and this will help you learn to adjust your listening for your monitors.

2. The bass will typically be under-represented on small studio monitors. That doesn't mean you should add bass however. You should simply understand their shortcomings and look for ways to get a good perspective on how much bass there is in your mix and master.

HEADPHONES

Headphones are another option for monitoring. There are entire sites and forums dedicated to headphones (such as **http://www.headphone.com**) so again we'll leave our hardware recommendations out of it and just advise you to do your own research. When working with headphones, here are a few things to keep in mind:

1. Imaging on headphones is very different than imaging on speakers. Usually music produced on a well-imaged speaker system will work well on headphones. The opposite is not as reliable, so use headphones as an alternate perspective, but not as your main monitoring system if you can avoid it.

2. Equalization can be very different on headphones compared to loudspeakers. The listening room, your head and even your outer ear have filtering properties that alter the frequency response of the music. This "natural equalization" is bypassed when you listen on headphones. If you're interested in learning more about this phenomenon, look into "diffuse field" headphones.

APPENDIX B: GENERAL OZONE TOOLS

We've touched on some of these during the course of explaining different modules and effects, but here is a summary of the tools available to you in Ozone, in addition to the mastering effects themselves.

MULTIBAND EFFECTS

A standard compressor or stereo widener can be a useful tool for processing your mix. The possibilities become even more interesting when you're working with multiband effects. With multiband effects, you can apply processing to individual bands or frequency regions of the mix. This means that you can choose to compress just the dynamics of the bass region of a mix, or just widen the stereo image of the midrange.

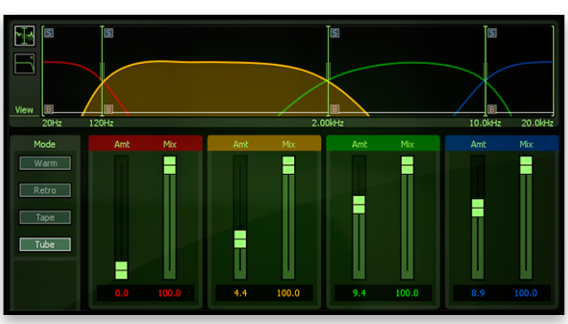
Ozone includes three multiband effects: A Dynamics processor, a multiband Stereo Imaging control, and a multiband Harmonic Exciter. To get the most out of these effects, it's useful to first take a moment to consider the multiband concept and how to setup multiband cutoffs for your mix.

Multiband effects have been around for many years in hardware. Engineers realized long ago that they could filter the bass of a mix with an equalizer, route the filtered output of the equalizer through a compressor, and then mix the output of the compressor back into the mix. Software plug-ins eliminate a lot of the wiring complexities of using multiband effects, but still present design challenges of their own. A multiband effect is essentially splitting your mix into frequency regions, processing them independently, and then combining them back together again. In order to sound natural, the design must carefully compensate for how the bands are split apart and recombined.

Ozone has been developed to perform multiband processing with extremely tight phase coherence, which means that you have the power of multiband processing while retaining a natural transparent sound.

Using Multiband Effects in Ozone

Before diving into the effects themselves, the first step is to listen to your mix and determine where to set the band crossover points. Load up a mix and switch to one of the multiband modules (the Harmonic Exciter, for example):



At the top of the screen you can see a spectrum divided into four bands. The vertical lines represent the crossover points of the multiband effects.

You can adjust the band cutoffs by clicking and dragging them with a mouse. You can also use the arrow keys after selecting a band cutoff (click the handles near the center of each cutoff line to select).

Setting Multiband Cutoffs

So where do you set the bands? In general, you want to try to split your mix so that each region captures a prominent section of your mix. For example, the strategy behind the default band cutoffs is as follows:

> **Band 1:** This band is set from 20 to 120 Hz, to focus on the "meat" of the bass instruments and kick drum.
>
> **Band 2:** Band 2 extends from 120 Hz to 2.00 kHz. This region usually represents the fundamentals of the vocals and most midrange instruments, and can represent the "warmth" region of the mix.
>
> **Band 3:** Band 3 extends from 2.00 kHz to 10 kHz, which usually can contain the cymbals, upper harmonics of instruments, and the sibilance or "sss" sounds from vocals. This is the region that people usually hear as "treble."
>
> **Band 4:** Band 4 is the absolute upper frequency range, extending from 10 kHz to 20 kHz. This is usually perceived as "air."

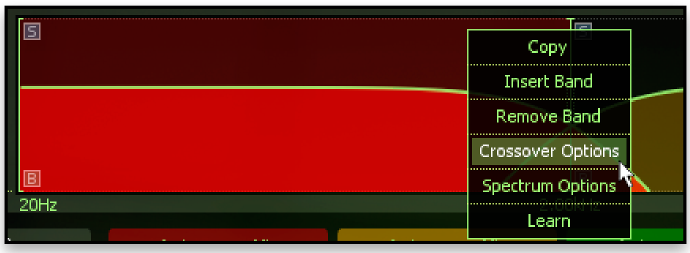
Keeping in mind that instruments have harmonics that can extend over several octaves, the goal is to try to partition your mix into bands. Play your mix, and click on the "S" button on each of the bands. This solos the output of that band. Now that you can hear exactly which frequencies are contained in each band, try adjusting the band cutoffs by dragging them with the mouse.

Note that you can choose to use 1, 2, 3 or 4 bands – i.e. you don't have to split up your multiband processing into four bands. Less bands can be easier to manage in some cases, and can also conserve CPU processing. To add or remove bands, right-click on the mini-spectrum as shown below and choose "Insert Band" or "Remove Band."

Crossover Options

As mentioned before, the first step of multiband processing is to split up (filtering) the signal into different bands. As such, you have the option of using analog filters or digital (linear phase) filters for multiband effects.

To select the crossover type, right-click on the mini-spectrum and select the "Crossover Options" menu item as shown below.



Once selected, the Crossover Options screen appears. Here you can specify whether to use analog or digital filter models for the crossovers. Note that if you use digital crossover, you can also specify the slope or "Q" of the crossovers.

Which should you use? As always, it's a matter of taste. Analog multibands can provide a desirable coloration of the sound, while digital crossovers provide linear phase "transparent" processing. The Hybrid crossover is the perfect reconstruction IIR (infinite impulse response) analog crossover. It's designed to reduce phase distortion and frequency distortion found in other analog crossovers while maintaining precise crossover points and the warm characteristics of analog crossovers.

Multiband Main Points

If you can hear the "parts" of your mix captured in each of the bands, you're in good shape. If you don't know exactly where to set them, don't worry. Once you start applying processing to each of the bands you'll begin to develop an intuition for where they should be set. The main ideas at this point are simply:

- Multiband effects are applied independently on up to four separate bands.
- Each band should represent a musical region of your mix (bass, warmth/vocals, cymbals/harmonics, air, etc.).
- You can adjust the cutoffs of each of these bands.
- You can mute the output of the bands to hear exactly what is passing through the remaining bands (Alt-click the "S" button).

- So let's just leave it at that for now and have some fun with a little multiband processing.

OZONE TIP Right-click on any control and you can copy its value to the clipboard. From there, you can paste the value to another control, or even another text application (Notepad, Excel, etc.)

MID/SIDE PROCESSING

At the mastering stage, Mid/Side processing separates an ordinary stereo recording into its separate areas of your soundstage independently. Mid/Side Processing lets you work independently with the center of the stereo image or the edges.

What can I do with it?

You can surgically EQ the mud out of hard-panned guitars while preserving the vocal and kick drum in the center of your mix. You can add excitation or a touch of room ambience to just the edges. Ozone's EQ, Reverb, Dynamics and Harmonic Exciter modules are all capable of Mid/Side processing, so (especially with multiband modules) the possibilities are endless. Again, experiment!

Standard Mid/Side Controls

Stereo | Mid/Side: This button allows you to switch between "Stereo" processing or the new "Mid/Side" processing. When "stereo" is selected, the Ozone module will perform basic stereo signal processing. When "Mid/Side" is selected, all Mid/Side features become available. Note that when the audio file is mono, the "Stereo" button will be labeled "Mono" and Mid/Side controls will be disabled.

Mid: When selected, processing is applied to the center of your soundstage. The more you boost the Mid channel, the more centered (mono) the audio will be.

Side: When selected, processing is applied to the edges of your soundstage. The more you boost the Side channel, the more "spacious" the audio will be (remember, a little goes a LONG way).

Mid/Side Link: When selected, this allows you to make changes to both the Mid and Side channels at the same time. This tool helps improve workflow. By making changes to both channels simultaneously you don't need to switch back and forth between modes.

Different Modules with Mid/Side Processing

Mid/Side Processing is supported by almost all of the modules in Ozone, including the EQ, Reverb, Harmonic Exciter, and Dynamics. Throughout each of the modules, controls and meters will be color-coded with orange and blue representing the Mid and the Side of the stereo mix, respectively.

OZONE TIP

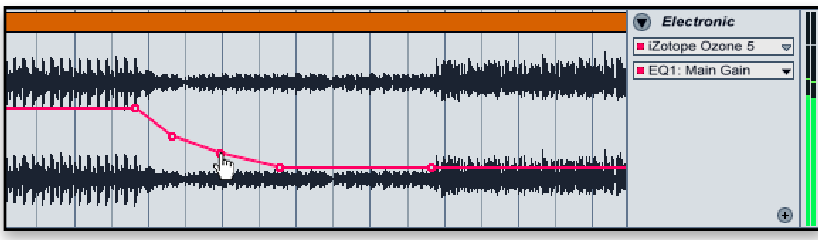
See Ozone's help file for specific information regarding Mid/Side Processing and how it works within the individual modules.

AUTOMATION

You can automate more than 370 parameters in Ozone using host apps that support automation, such as recent versions of Avid Pro Tools, Sony Sound Forge, Cakewalk SONAR, and Ableton Live. Automation allows you to specify changes to parameters over the duration of a mix—such as stereo widening during a chorus or boosting an EQ during a solo.

The implementation and specifics of automation are dependent on the host application, so we refer you to the documentation of the host app for setting up an automated mix with plug-in effects. In general, though, you patch Ozone as an ordinary effect on a track, then in the track view of the host app assign automation envelopes to it. These envelopes control how Ozone parameters are changed over the course of the mix. In this case, most of your "tweaking" is done in the track view of the host app, dragging or drawing curves and envelopes as opposed to changing controls yourself in Ozone. For example, in the screenshot below we're adjusting the gain of EQ Band 1 in Ozone from the Arrangement View in Ableton Live 8.

When automating in a track view with envelopes, but working mainly with the Ozone interface, we find it helpful to be able to "see through" Ozone



so you can monitor Ozone meters and controls but see the track view and automation curves behind Ozone. So we provide an Opacity slider in the main options dialog. This allows you to see through Ozone to monitor both what Ozone is doing and what is happening with the automation curves. Note that only certain hosts allow this feature in plug-ins.

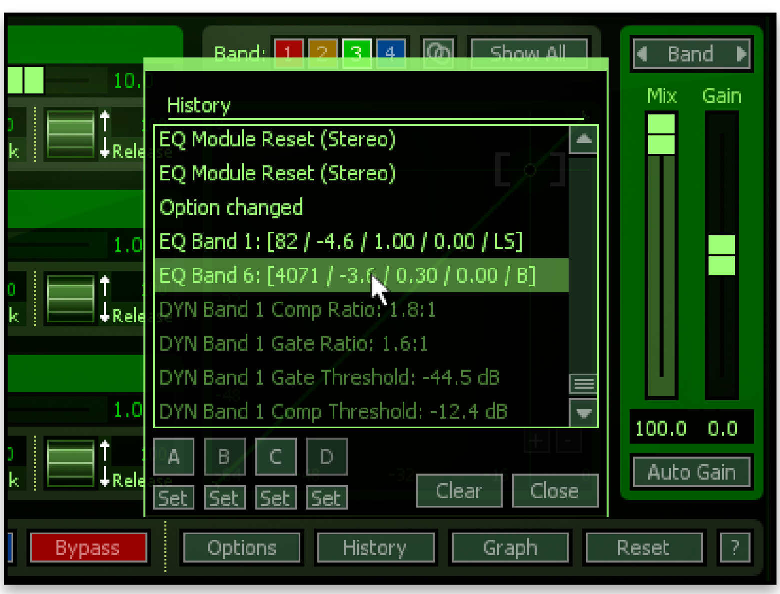
MASTERING QUICK TIPS

- Try using automation on stereo widening. Widen a chorus and tighten a verse for starters.
- Try changing compression ratios between choruses (hard/high ratio) and verses (soft/low ratio).
- Try increasing the Exciter amount on light/quiet parts. Back off on the amount during louder/fuller parts.

HISTORY LIST

As you tweak controls, each movement is captured and displayed in the History list. To go back and hear a previous setting, simply scroll up the list and click on the point you want to audition. When working with mastering effects, perhaps most notably EQ and dynamics, it's very easy to get accustomed to the "new" sound and overdo the effects. The History list lets you quickly jump back for comparisons to refresh your "auditory memory."

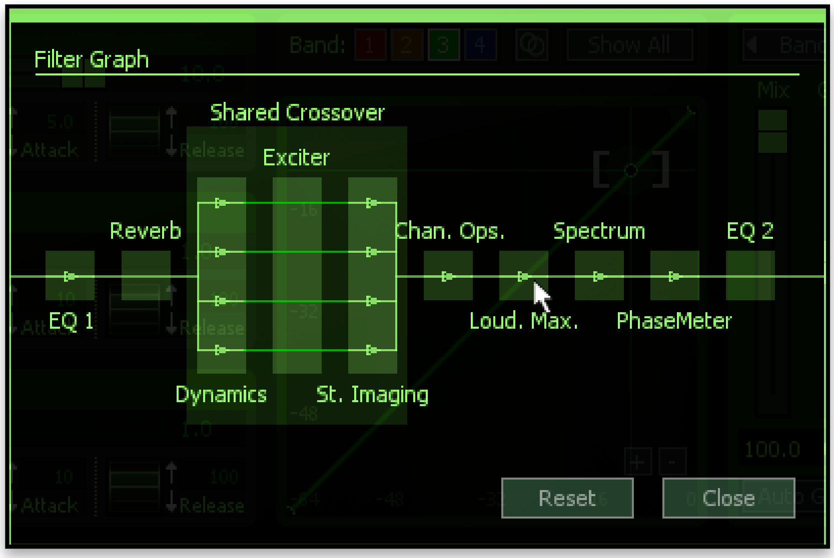
There are also times when you need to decide whether to use (for example) the Harmonic Exciter or a high frequency EQ band to bring out the upper end of the mix. Try one way, then try another way, and click back and forth on the History list to compare. If you want, you can assign a specific point to the A, B, C or D buttons for even quicker comparisons. The History list and any settings assigned to the A, B, C or D buttons will be cleared when you unload Ozone. To permanently save a setting, save it as a preset (to be explained shortly).



SETTING THE ORDER OF THE MASTERING MODULES

By default the order of processing in Ozone is the order listed below (the signal passes through the EQ, then the Reverb, etc.). The phase and spectrum meters are based on the final output signal (after all processing, except the Post Equalizer which is at the end by default):

1. Equalizer
2. Reverb
3. Dynamics
4. Harmonic Exciter
5. Stereo Imaging
6. Maximizer
7. Post Equalizer



You may simply prefer the sound of a different order, or you might want to reorder just for a specific purpose. If so, you can change the ordering of the processing by clicking on the Graph button. This provides a signal path flowchart of the current settings. You can modify the order by selecting a module and dragging it with the mouse to a new position. You can also place the spectrum and phase meters at any place within the processing if, for example, you'd prefer to see a spectrum of the audio going into the EQ as opposed to seeing the spectrum after equalization.

The Shared Crossover block actually represents the three multiband modules: Harmonic Exciter, Stereo Imaging, and Dynamics. For convenience, you can reorder the position of the multiband modules in the signal chain by dragging the entire single block labeled "Shared Crossover." You can also reorder the individual multiband modules within the block, or even move them in between other single-band modules outside of the "Shared Crossover" block.

The new signal order is applied immediately when you drag and drop a module. If you want to reset to the default order, click the Reset button.

Note: While it is possible to reorder the modules while audio is playing, keep in mind that a different signal order can create completely different sounds and levels, even with the same settings for each module. Reordering while audio is playing could cause a sharp jump in output level. Stop the audio from playing, reorder the modules, and start the music playing again with one hand on the level of your mixer.

PRESET MANAGER

Ozone includes a complete system for working with Presets. To access the system, click the Presets button on the Ozone faceplate.

Some of the benefits of this preset system include:

- Easily backup and transfer preset files.
- Store preset files in folders for easy access and management.
- Undock the Presets window for easy access at any time.
- Share Ozone presets across multiple host applications.
- Sort presets based on name, date last modified or date last used.
- Update presets with one click.

- Selectively load settings from individual effects modules. For example, take the EQ settings from one preset and combine them with the dynamics settings of another preset.

- Add comments to presets for easy reference.

To learn more about using the Preset Manager, we invite you to check out this topic in the Ozone help file (like always, click on the "?" button, this time in the Preset Manager window, to access the Ozone help file).

SHORTCUT KEYS AND MOUSE WHEEL SUPPORT

Here are some shortcuts worth highlighting as you get started with Ozone:

Wheel Mouse

- If you have a wheel mouse, you can adjust most controls by simply positioning the mouse cursor over the control and rolling the wheel. Some controls require clicking on them first, and then rolling the wheel to adjust them.

- In the Equalizer, you can adjust the Q of a selected band or bands with the wheel after clicking on the one you want to adjust.

- In the Paragraphic Equalizer table (accessed with the "Show Info" button) you can adjust a value by holding the mouse over the value and rolling the wheel.

- In the History screen, you can use the wheel to scroll through the History list.

Numeric Entry

Using the mouse can be difficult for accurate slider settings. Here's how you can adjust values with precision:

- You can select the slider you want to adjust by clicking on it. If you click on the slider itself the value will jump to the mouse position, but if you just click on the label of the slider, the slider will be selected without changing the value.

- Selection or "focus" of a slider is indicated by white brackets around the slider. Note that by default the focus bracket "travels" from slider to slider, but you can turn this off by unselecting "Enable animated focus" in the Options dialog.

Once a slider is selected, you can use:

- Arrow keys to move the slider in small amounts.

- PageUp and PageDn (or Shift+arrow keys) to move the slider in medium amounts.

- You can also enter a numeric value directly. Double-click on a slider's numeric value, and a manual entry box should appear, allowing you to type in a new value. When you're done, just press "Enter" to close the box.

- You can Undo or Redo your last change using Ctrl+Z and Ctrl+Y.

Meters

- You can zoom in and out on certain meters by holding down the Ctrl key (under Windows) or the Command key (under OS X) and clicking with the left mouse button to zoom in or the right mouse button to zoom out. Under OS X, you can zoom out with a Ctrl-Command-click.

- You can reset the peaks or averaging of the spectrum by clicking on the spectrum.

- You can reset the peak indication of the level meters by clicking on the meter, or reset the clipping indicator by clicking on the clipping indicator.

- You can reset the peak hold of the Vectorscope and phase meter by clicking on the meter.

Options

Don't be afraid to right-click (under OS X, you can also Ctrl-click) on things in Ozone. In most cases (meters, I/O gain controls, spectrum, etc.), a right-click will present you with an options menu for that object. So, for example, if you right-click on the I/O level meters you can access options for the I/O meters. Right-click on the spectrum and you'll get Spectrum Options. And so on… And remember, if you aren't using a two-button mouse under OS X, you can just substitute a Ctrl-click for a right-click.

OZONE TIP

For a complete table of shortcut keys, and for other in-depth information about using Ozone, be sure to check out the Ozone help file. In the meantime, thanks again for downloading and reading our guide to mastering with Ozone!

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