Ozone 9 Advanced Partial Cheat Sheet

The script Adam from Real Home Recording used for the Ozone 9 tutorials series.

User since version 3, used the demo on the first album I ever mixed. Bought the boxed version 4 which also had a free version 5 upgrade. A few months ago IK sent me an Advanced license to try out. So, to say this video series has been delayed would be an understatement.

iZotope's Ozone 1 was released 17 years ago. Version 3 was my introduction to it. I have used every iteration of it since then except 8.

Ozone is a suite of audio processors that are intended for music mastering but they can be used during mixing too. For this video series, I am using version 9.1.

The Ozone 9 suite features multiple equalizers types, a few different compressors, an excellent brickwall limiter, tape emulation, mix referencing, lossy codec preview, an excellent dither and much more. A standalone application is also included.

New stuff:

Standard got a lot of new features that Advanced 7 and 8 had.
Advanced got upgraded Master Assistant, Tonal Balance Control and Low End Focus modules.

Both versions get Match EQ, which was an old removed feature from the Ozone 5 days but apparently this one works better.

Resizable GUI but more on that later.

Extra module slots...so you have to load up a second instance of Ozone a lot less.

Ozone can be an overwhelming plugin at first. I will do my best to break it down so you can use it to its full potential. Most of my information comes directly from the user manual. Once you learn the main plugin, also known as the Ozone 9 Mothership, the individual component plugins are easy.

Before we get started, make sure to set your DAW/audio interface buffer size to a higher one. I usually select 1024 but for larger mixes 2048 or higher can be helpful.

Four Main Areas of the Mothership plugin

Top Part - Called the Global header.
Inter Plugin Communication name editor, Master Assistant preset manager,

undo, undo history, settings and help documentation.

The undo history's set control is very handy!

2. Signal Chain. Lets you see an overview of the modules and adjust their order.

3. Module Interface. Here is where the unique controls are for the different processors within the Ozone suite. Modules can only be added once except the equalizer.

Some modules can be processed in different ways. Stereo, Mid/side or for the equalizers independent left and right channel processing. Look for the double rings icon.
Select Mid-side or left right reveals more options.

The neat thing about the parameter linking is that they are relative.

Meaning, if you adjust a control on one then click the Link button, if you adjust the control further it will only move a little more. it won't synch up to the previous control.

4. Input/Output Panel. Includes meters, global gain controls, channel operations including swap, mono summing and more.

Clicking the I/O button gives you some settings.
Input and output gain link button is at the bottom of each meter. Allows for individual left/right controls. You can unlock, adjust then re-lock to create an offset.
+/- button next to those changes the meter scale.

Click and drag the sliders up and down or use your mouse wheel after clicking a slider.
Double click sliders to reset them.

Underneath the I/O panel are audition controls.
Bypass is a global processing disable function.

Gain match goes hand in hand with the bypass button. it will level match the unprocessed signal to the processed signal. Keep it on.

Mono sum button is helpful if your DAW doesn't have one. Left and right channels and turned into one.

Left/Right channel swap, same thing. The left signal is sent to the right side channel and vice-versa.

Reference button opens up the Reference tool where you can load different audio files for easy A/B comparisons. I'll go into details about this tool later.

Codec button is the Codec Preview tool. If you want to hear what your music will sound like when it's being data compressed such as with music streaming services, use this. The solo artifacts button and preview buttons make this process a breeze.

Bottom right is the resize drag.

The input gain control is first in the signal chain. The input meter is next followed by the first processing module.If the maximizer or vintage limiter modules are used, the DC offset filter is applied before those if it is enabled.

After the final module, the output gain is in the signal chain followed by Dither and the Codec Preview tool. The output meter is the last one.

Component plugins allows CPU resource savings. They are also easier to automate.

Total Balance Control 2

Metering plugin included with Advanced. Displays spectrum information for a track relative to a target curve. I won't be talking about this plugin within the Ozone 9 video series.

When I first started using Ozone over a decade ago, the first thing I did was click through the Presets. This gave me a general idea of what the plugin was all about. I suggest doing this for newbies not just to learn from but also to use as a starting point.

iZotope have included a large number of presets both in the mothership and component plugins. The Master Assistant is another excellent starting point and the method I personally recommend first once you take a listen through the presets.

Assistant Settings:

Modern modules are the clean/digital variety.
Vintage modules are great for that old school vibe.

Plug a few options into the Master Assistant and then cross your fingers.

You can either have it do its thing automatically or you can choose a target reference. That could be a mastered song from the album that you are already happy with or a commercial release that you or the musicians you are working to enjoy.

Loudness. A very important setting. Manual gives you three options which are based on the artificial intelligence's genre analysis. Low is -14 LUFS,

Medium is -12 LUFS and high intensity is -11 LUFS.

You can also give it a Reference track, so the loudness target is based off that.

Destination sets the maximum peak to either -1.0 dBFS for streaming or -0.3 dBFS on the CD setting.

Play it back on the loudest portion of the track for at least 30 seconds.

You can also loop if the section isn't more than 30 seconds. If you don't do this, the loudness target will be inaccurate.

Don't be surprised if the Master Assistant disables the Dynamics module.

If you track is louder than the target loudness, it will not be made quieter than the ceiling value.

CPU saving: Adjust buffer sizes in options menu. Disable meters. Do not tweak anything unless necessary.

The icons are important to memorize. The + sign is the add button. Power buttons enable and disable bypass.

S solos the module.

The three dots and lines icon is the module presets menu.

X removes the module. If you click it by accident, click the upper right undo button.

Six dots allows you to click and drag to change the module's signal flow order.

Each module also has tiny meters. The left one is the input signal and the right side is the output level after the module's processing. No module gain controls exist apparently though...

Back to the I/O Panel, click the I/O button to display some settings. Type, Source and two boxes are interesting options.

Descriptions for the metering types are on screen.

Source options. Mid/Side shows three meters. The center one is the mid channel and left/right are the sides. The gain sliders are not affected by this option.

Module Views

Options to change the appearance of plugins is on the upper left side in the Module interface area. Depending on the module, it may have some or none of these options.

All Bands - Controls on bottom

Correlation Trace - A scrolling history of the stereo correlation. Light blue = in phase and red is out of phase.

Crossover Spectrum - View processing bands in the multiband modules. You can double click the handles to type in crossover frequencies. Add bands in that little gap underneath the View icons.

Detailed Band - Default EQ module view. Basic parameters are shown with a pop up heads up display on the bands.

Detection Filter - Dynamics and Vintage Compressor modules. See the detection circuit's frequency response.

Gain Reduction Trace - Shows the waveform history for compression modules.

In the Dynamics module, only the selected band is displayed.

Post Filter - Displays saturation. Affected frequencies are white. Non-affected aren't shown.

Spectrum Analyzer - Displays a real time frequency analyzer of the module's output.

Stereo Width Spectrum - Self explainatory.

Ozone 9 Standalone application benefits:

Get outside of your DAW. Little bit psychological but it's there.

Apparently the Ozone session file includes imported audio tracks.

Control - i to import files or click the plus sign. Drag and drop also works. Up to 16 tracks are supported.

Control - E is export

Limited file format and sample rate support. Says it doesn't accept 32-bit Float point files?

MP3 Codec: LAME
AAC Codec: Fraunhofer
Only 44.1 kHz is supported. No Opus support.

If you import a second file, its sample rate will be converted to that of the first imported file.
No manual control over sample rate conversion, unlike in iZotope RX.
Why this still is the case is baffling to me. I guess to keep things simple.

Standalone app allows for external plugins.

It's unclear if VST 3 is supported. In the manual, it may confuse people to not say that AU is Mac only

Dynamic EQ

It is not unusual for portions of the frequency spectrum to stick out too much during the course of a song. Dynamic equalizers tame these frequencies.

To make this tutorial easier, I'll go across the top, left to right and then down when showing the module controls.

Filter ModeAnalog: minimum phase IIR filter types.

Digital: Linear phase FIR filter types. Uses more CPU, retains the original signal's phase but could have pre-ringing artifacts.

Channel Processing Mode. Stereo, M/S and Left/right

Factory Reset button. If you accidentally click it, use the undo button (upper right).

On the left is your Spectrum Magnitude Scale. When your audio is playing back, you'll see the real time frequency analyzer in the background

In the middle we have our four band node points. To add a new one, double click anywhere inside the spectrum area. Depending what portion of the frequency spectrum you add a node will determine the default filter shape.

Far left = Flat highpass or Low Shelf
Far right = Flat lowpass or High Shelf
Middle: Proportional Q

Pass nodes are only added once.

Clicking any node will reveal a pop up box with more controls. A bypass button, solo and remove band button on the left. At the top is a filter selector menu with four choices.

Proportional Q: As you boost or cut, the Q factor AKA bandwidth changes.

Similar to an API equalizer. Unlike an API, you can also alter the Q factor.

Bell: A standard parametric equalizer peak bell filter. Bandwidth does not change as the gain changes.
Band Shelf: Flat top EQ. Useful for boosting or cutting a block of frequencies.
Baxandall Bass: Transparent/gentle boosts and cuts.
Bax Treble: Same as the Baxandall bass but for the high frequency ranges.

To adjust the center frequency or in the case of pass filters, you can drag the node left and right, enter the value by double clicking it then typing or using the left/right keyboard arrows.

To change Gain, drag up or down, manually enter the value or use up/down keyboard arrows.

More tips: Hold the shift key to lock horizontal frequency movement while dragging left and right. If you click the node and then hold shift, you can drag up and down to change gain without changing the frequency.

If you hold Control while using the arrow keys, more precise adjustments can be made. Unless you're using REAPER. Shift + arrows = greater adjustments.

Double clicking a node resets its parameters except frequency.

For Q, you can use the mousewheel to make it narrow or wider (higher numbers are narrower) or manually enter it. Some DAWs can use the Alt + keyboard arrows shortcut, REAPER can't.

There are dots to the left and right of the node. You can drag these left/right to adjust Q as well.

Now, for the Dynamic EQ section of the parameters. The Threshold point determines when gain adjustments will occur. If the slider is all the way up, no gain changes will occur. You can either drag it or the more precise way is to click the slider then use keyboard up/down arrows. You can also

double click to enter a value.

The gray meters show the input signal. The red meter bar shows gain reduction.

Ready to get a little more complicated? Above and below the node are arrows. These control the Dynamic Mode direction and are Down by default. In the down mode, if a negative gain value is used then that band will be reduced when trigged. If the node has a positive gain then it will not be boosted as much if a signal exceeds the threshold.

The idea is to set a maximum band reduction point and then only when those frequency exceed a certain level will a cut take place. Just the same, you can boost a certain frequency range but if it gets too loud then the boost is reduced slightly or altogether.

In the Up direction, which I myself will rarely use, if the node's gain is positive then a frequency boost will occur when the threshold point is triggered. If the node gain is a negative value, a cut when happen.

Finally, Click the arrow on the right of the Threshold area to open up the

Advanced Panel. By default it's set to Auto Scale. If you want to manually set Attack, Release and Offset values. you can in this box.
It works similar to a compressor.

In addition to the Solo button, you can hold the alt key to click and hear just that node's area. This is called Band Solo. If you change the Q factor/bandwidth it does not update. You have to move the gain up or down a little to hear changes.

Alt Solo allows you to hear an area outside of the node.

To remove nodes, select and click X. On some DAWs, you may be able to select multiple nodes and use delete or backspace to remove them. It didn't work for me in REAPER.

Composite Curve: The gray line. Combines all of the enabled bands and displays the filter response shape. Changes while the track is playing if a band is triggered.

Click anywhere on the curve to add a node point.

Filter Response Curve: Displays the shape of the currently selected band node. It will be a different color depending on the selected band. If a band is bypassed, it is gray. Multiple nodes can be selected with a control-click. If no nodes are selected, the FRC is hidden.

Spectrum options can be found under the options menu.

On the right is the Dynamic EQ gain scale.

At the bottom is the Spectrum Frequency Scale. It's an easy way to see where to put node points.

Hold shift to move the node frequency left to right without changing gain.

You can also select multiple nodes by clicking and dragging them with a rectangle box.

Dynamic EQ Options menu

Musical Units: Music notes instead of frequency points at the bottom.

Alt-Solo Filter Q: Default bandwidth setting.

----Dynamics module -
Multi-band compressor/expander. Up to four bands.

Similar to the Dynamic EQ except it uses cross overs. I would recommend watching that tutorial and the Ozone 9 Mothership overview first. If you didn't, that's OK. I will briefly go over controls I've already talked about.

Controls wise I will go left to right and top to botom.

Views Selector
Crossover
Gain Reduction trace
Detection Filter

Channel Processing Modes: Stereo or Mid-side
M/S is helpful if you want to focus processing on vocals.

Add crossovers:

In Crossover View, hover over the top area. A plus sign will appear. Click to add a crossover point.

Learn Button: Cannot be used until crossovers are added. Be in Crossover view and playing the track. It will shut itself off after natural crossover cutoffs are defined.

Reset button: All controls back to the default values. If you accidentally click it, use Undo History.

Each crossover has its own set of compressor and limiter controls. The compression proceeds limiting, signal flow wise.

Ratio, attack, release and knee controls are available along with of course

Threshold. If you don't know how to use a compressor, check out my in depth tutorial video which I'll link below and in the upper right corner.

Double click the numbers to manually enter values.
Keyboard arrows work after clicking a control.

Level Detection Modes
This setting will determine how compression is applied.

Peak: Useful to even out spikes in audio
Env: Envelope mode. A more advanced and probably better sounding RMS Mode.
RMS: Old school average level detection. Can create unwanted artifacts but is more natural sounding than Peak.

If you set the Ratio to negative values, it becomes an expander.

It should be noted, even at the maximum ratio, overshoots are still possible. This plugin isn't a brickwall limiter.

Wet/Dry Slider: At 100%, only the processed signal is audible. At 50%, the unprocessed and processed signals are split evenly. Allows blending with the uncompressed signal for what's called Parallel Processing.

Dynamic Curve Controls:
Another way to modify the Limiter and Compressor's Threshold and Ratio controls.

Bottom right, zoom in and out of the Dynamic Curve meter.

Next to the Parallel slider is the Bands view. You can choose single band or all bands. In single band mode, arrows allow easy cycling between the different active bands.

Want to adjust all the crossover band controls at once? Enable the Link

Bands mode. Under the Single/All Bands view buttons.

Adaptive Release and Auto Gain are interesting features.
Adaptive release works in relation to the manual Release value.

Dynamics Settings menu:
These options have a pretty decent effect on the dynamics module. They also effect the buffer needed on the plugin.

You can pick analog, digital or hybrid, crossover types. Listen while switching to hear the difference. Analog sounds natural but with artifacts, digital more transparent. Hybrid is the best of both worlds.

With digital, the buffer size and Q options open up.

Look Ahead can affect how transparent the module is, at the expense of a larger buffer. The higher, the more transparent.

Exciter

One of the easier to use modules. This could be called the Saturation module, because that's what it does. Encourage viewers to watch the Ozone 9 playlist.

Views: Crossover - The Ozone 9 multi-band view. Allows insertion of crossover points. Also has a frequency spectrum graph. Top part lets you add, number is displayed below. In between the vertical bars, you can remove the crossover, bypass, Solo.

If no crossover points are added, the entire frequency range is saturated.

Post Filter: Displays how much saturation is being applied across the frequency spectrum. Indicated in white. The big white dot is a low pass filter option that can tame harsh frequencies.

Channel Processing Modes: Stereo or mid-side. Processing upper frequencies on the sides could benefit a mix.

Learn button: Add crossovers and it activates. Searches for natural crossover cutoff points. Auto disables.

Reset: Factory default module settings.

Oversampling: Pumps the sample rate up to reduce aliasing artifacts. It's either on or off for all Exciter bands. Requires a little extra CPU usage.

At 88.2 khz or above sample rates, it may not be necessary.

Link Bands: Controls are linked for all bands.

Modes
Analog: Odd harmonics, transistor.
Retro: Different type of transistor, odd harmonics.
Tape: Brighter saturation, characteristic of a tape machine.

Tube: Clear excitement, emphasizes transients
Warm: Quick decaying even harmonics
Triode: Half of a Tube circuit. Warm and subtle.
Dual Triode: Full tube circuit. More pronounced overdrive and warmer tone.

Amount: Self explainatory.
Mix: Blends the unprocessed signal with the saturated signal.

Exciter Settings/Options menu:

Once again, crossover options like the other multi-band modules.

Imager

Adjust the stereo width of a mix. Also provides nice stereo meters to ensure mono compatibility.

Controls are set up similar to the Exciter.

Views
Crossover Spectrum
Correlation Trace
Stereo Spectrum

Learn - Auto find crossover points. Enable bands first. Automatically stops.

Reset

In the Meters and Displays area, when in Crossover Spectrum mode, you can add bands, solo them, remove them or bypass. Plus of course move crossover points around.

Stereo Width Sliders - Positive numbers are wider. Negative are narrower.

At -100, that frequency band is mono. Move them up and down with the mouse or double click the number and enter a value between -100 and 100.

Link Bands button

The two other stars of the show are the Stereoize modes. Mode I is the classic Haas Effect which has been in the Ozone since version 5.

Mode II is the new mode called Velvet Noise decorrelation. It preserves

transients at high slider settings.

The Vectorscope shows Ozone 9's output, no matter where Imager is in the chain. Click on it to reset. Reading this is a tutorial video in itself.

There are three metering modes:

Polar Sample: Dots are plotted per sample. Patterns that fall outside of the safe lines are out of phase.

Polar Level: Displays rays instead of dots. Audio inside the safe lines are

in phase. Ray angle shows where the are in the stereo image.

Lissajous - A fancy name but it's similar to the Polar Sample scope except it's shaped like a diamond. Vertical patterns are good, horizontal patterns aren't.

Samples that are clipped turn red.

Phase Correlation Meter: Measures how out of phase a track is. If it is pinned at +1 then it's mono. Don't let it drift under 0 for more than a second or so. When mixing, solo tracks to meter best.

When setting the band's stereo width setting on a mix during mastering, pay attention to this meter the most.

My advice? Watch these meters while listening to commercial music. See how they look.

Imager Options menu:

Prevent anti-phase checked for complete mono compatibility.
Processor may not sound as wide though.

Vectorscope options: Best left on the default peak. Allows us to see the maximum signal level.

Crossover type: Just like the Dynamics tab options. Digital mode offers the most precision/transparency. Play your track, adjust these settings.

Spectral Shaper

Similar to dynamic EQ and multiband as well, except easier to use. The main difference is the Tone control.
Better used on tracks. Similar to a de-esser.
'
Interface is similar to Low End Focus.

Views, upper left.
Channel Processing
Reset

Adjust action region with the Range Selectors.

Solo the region, pre-processing.

Intensity Modes on the left. Self explanatory.

Threshold seems to be broken. Gain reduction meter looks accurate, but don't set the level while looking at the input levels.

The Listen button allows you hear what the processed audio is doing.

Tone: Sets the spectral tilt of the processor. Positive numbers are brighter and negative numbers are darker.

Attack/Release: Just like compressor controls. Attack is how long it takes for the shaper to apply gain reduction. Release is how long it takes for the shaper to stop applying gain reduction to signals that fall below the Threshold.

Vintage Compressor

Feedback compressor emulation. Unlike modern compressors, the level detection uccurs at the output.

By default, the Detection Filter view is shown. You can either leave these node points as-is or adjust them for different effect. On the far left is a high pass filter, Middle node is a wide Q parametric filter. The final node point is a high shelving filter. Use the solo button on the far left to hear what's being fed to the sidechain.

Modes:
Sharp emphasizes transients so your signal sounds crisp.
Smooth Makes transients sound duller. The result is a fuller/thicker sound.
Balanced is a compromise between these two modes. Dynamics get preserved and the signal body is enhanced as well.

If you are familiar with how a compressor works, the rest of the controls are a breeze. A quick overview is, Threshold sets the level at which gain reduction occurs. Ratio is how much gain reduction occurs. Attack is how fast the gain reduction occurs and release is how fast gain reduction stops occuring after the signal goes below the threshold.

Auto Gain, on by default, compensates for gain reduction so the volume level doesn't change so drastically.

Vintage Equalizer

An approximation of the Pultec EQP-1A and Pultec MEQ-5. Top row of controls is the EQP-1A and bottom is the MEQ-5.

I'm not a really big fan of this module. Mainly because with the real hardware or other plugin emulations, you can boost the high end and it basically never sounds bad. That's not the case with this one.

At least the low end boost trick is in tact but again, it has a digital sound to it.

Vintage Limiter

Inspired from the Fairchild 670. Want a louder and full master? Use this module.
A good processor to put before the Maximizer or as an alternative to it.

Modes:
Analog has a fast attack and program dependent release time. Smooth with a thick low end.

Tube showcases variable attack and release times. Prevents clipping, despite its non-linear analog nature.

Modern: Thick vintage character + modern brickwall limiting = this mode.

Threshold: Sets the point where gain reduction starts.

Ceiling: Like the Maximizer module, sets the maximum output level.

True Peak: If you need a very loud master, use this. Makes it so analog distortion does not occur.

Link: The threshold and ceiling controls become linked.

Character: Changes the attack and release times 10 is slowest, 0 is fastest.

Vintage Tape

Inspired by the two track Studer A810. Models the saturation and frequency response changes but not crosstalk, hiss, wow and flutter.

The presets are a great place to start! In the Mothership, they are on the bottom left of the module name box.

Choose tape speed based on how much fidelity you want to maintain.

Input Drive: Controls tape saturation level.
Bias: Shape of the distortion curve and high frequency response. negative numbers under bias the emulation, boosting high frequencies. Positive bias attenuates higher frequencies. At high numbers, dynamic range is severely reduced.

Harmonics: At 0.0, standard odd-harmonic tape saturation.Above zero, an out of spec tape machine emulation with even harmonic saturation becomes audible.
Low Emphasis: The head bump of the tape. Controls the low end resonant peak. At its default of 2, the souund is subtle. If you want more bass, up it. Less? Lower it. The tape speed affects where the resonance is. At 15 IPS, it’s at 100 Hz. At 30 IPS, it’s at 50 Hz. 7.5? 200 Hz.

High Emphasis: Compensates for high frequency loss. At 4.0, it is close to a flat high frequency response.

Codec Preview

Gives you an idea of how your audio will sound when converted to MP3 or AAC. So, you can compensate for the changes that may take place after lossy conversion. Not intended as the actual converter process...for that use the Ozone 9 standalone app.

Take note of output clipping that may occur. That’s why limiting to about -0.8 dBFS is suggested throughout the user manual.

Fraunhofer AAC and LAME MP3 are used.
The maximum sample rate is 48 kHz. Values other than that or 44.1 kHz will be converted in real time.

Artifact Solo is cool.

Presets Management

Global vs. Module Managers

Module presets travel between the mothership modules or component plugins

Three lines

The plus sign is the add button. Make your settings, click the + symbol and name it. Done. Simple. You can also add a comment for your preset which I love.

Delete Preset or Folder, Create Folder, Change Preset Folder Path lets you save your folder to a different hard drive location.

You can also update a preset with the floppy disk button.

Show at Startup. If you use presets a lot, check this box. It is on by default.

The Preset Window will not close until you press the Close button.

Accidentally clicked a preset and want to go back to the previous settings? Scroll up and click <Working Settings>

You can make a preset the new default by right-clicking it.

To reset the default preset to the factory settings, right click any preset and choose the option.

Mothership Options

Gear button

Options Tabs

Reset button is just for that tab, not the whole.

Enable analytics: Upload anonymous data on usage of the plugin/application to iZotope's servers.

If Ozone is giving you keyboard shortcut conflicts, you can change Keyboard Support to none.

The Spectrum Type Options are interesting. Play a track with an equalizer loaded while changing the options to see what they do

Linear is the standard frequency spectrum line that most digital equalizers display.

Full Octave and 1/3 octave split the bar graph spectrums at the octave points.

Critical is a nice feature. The frequency graphs are split up into areas that the human ear differentiates frequencies.

Peak hold is a feature that I'm surprised isn't on by default. It shows the maximum point of the frequency. A little unruly to look at in Linear mode.

Peak hold time is how long the peak hold is displayed.

Window Type is another "play a mastered track and watch how the spectrum graph changes" control. Most noticeable on the Linear type display.

The way I look to look at it is, watch an area. Does what you perceive in your mind match what you're seeing and hearing? If not, try a different Type.

Window Size: Use in conjunction with Windows Type. Time vs. frequency resolution. The higher the number, the greater the frequency resolution.

The default 4096 is usually a good choice.

Average Time: Real time shows near instant frequency display. Great for tracking down resonances like sibilance or when making EQ cuts. The slower averaging numbers allow you to see the track in a larger picture way.

When matching EQ across an EP or album during mastering, an average time of 5 or 10 seconds along with Critical spectrum type could prove helpful.

Frequency Scale controls the bottom part of the spectrum. How they are spread out.

MEL: Based on human perception.
Log: Logarithmic. Good for most EQing tasks because of the mid range width.

Extended Log: Gives more detail that favors the low and low mids.
Flat-Log Default setting. Gives a good balance to allows for easiest overall EQing.

I/O tab:

Meter Types:
RMS = old school analog Root Mean Square averaging
Peak = instantaneous sample value (when True Peak is selected) or analog waveform values.

RMS + Peak = Bright bar is RMS, dimmer but higher bar is the peak.

K-System: Bob Katz's loudness metering system.

Momentary: Loudness over 400 ms.

Short Term: Loudness over 3 seconds.

Integrated: Loudness over the course of an undefined period of time.

Short Term + Peak

Meter Scale: How the numbers are spread apart. dB (non-linear) is your best option for music mastering. Finer detail where it counts. 1771 and EBU are loudness scales. Especially useful when using meter zoom (+ and - buttons, center bottom).

Source let's you pick from Stereo or Mid-Side metering. In Mid-Side mode, the Mid signal is in the center. Left and right are side channel.

Peak Hold Time

Integration Time: for the RMS calculation.

Readout: Top meter number. Max Peak or current peak.

App specific options

Plug-in tab, enable/disable plugin formats.
Specify plugin folders.

Scan Plugins - Add newly added plugins

App tab

Segmentation overlay: When importing files, multi-colored segments are created by Ozone. It's a useful feature but if you don't want it, you can turn it off.

Playhead Follows Playback: When enabled, the playhead will stop where you pressed the stop button. This is off by default.

Ozone Equalizer

Controls are similar to the Dynamic EQ.
Two EQs can be used in the Mothership. Equalizer 1 and 2 are the same.

As in the other tutorials, let's start at the module's upper left. Global Ozone 9 controls were described in the Mothership video.

We have two Views. Detailed Band is the default and then All Bands.

Channel Processing Mode. Stereo, mid/side and Left/Right.

Reset button.

Explain how equalizers function.

Sides and bottom, meter scales. EQ gain on right, Spectrum Magnitude on left and Spectrum Frequency on the bottom.

Filter Response Curve
Shows what the currently selected node is doing. Represented by a colored line.

Composite Curve - Displays the response of all enabled bands.

New for version 9.1, Show Extra Curves. Displays how the equalizers affect phase.

EQ Filter Modes: Analog = minimum phase IIR.
Digital = linear phase FIR.
Analog and digital filters cannot be mixed.

Add nodes on the Composite Curve or by double clicking.
Remote Nodes: Click the X, click and drag then Delete key to delete multiple nodes

Left clicking a node will pop up the HUD menu. You can pick from 13 different Filter shapes, manually enter parameters, bypass or Solo the Band and delete it.

By default, the filter shape defaults to Proportional Q if added between

100 Hz and 8 khz. I'll talk about the shapes in a minute.

To move nodes:
Click and drag by the circle.
Click then use left/right arrows.
Manually enter values. Want 8 kHz? Type in 8k and so on.

Gain: Drag up or down or use up/down arrow keys.

Gain is limited to +6 dB or -10 dB unless you manually enter the values. At that point the max gain is +15 dB and -30 dB at the lowest.

Q: Handles, mousewheel or manual entry.

Band Solo: Alt and Click node. Q doesn't update.
Alt-Solo: Alt and click anywhere.

New for version 9.1, Text Scrubbing.
Simply hover your mouse over the frequency or gain HUD text, click then drag up or down. For fine adjustment, drag your mouse left and right. It kind of sort of works well on the frequency setting but not so much on the Gain.

Filter Shapes
Three main categories: Peak,

Peak: Allows for specific boosts or cuts at a given frequency.

Shelf: Choose a frequency and at that point all others past it will be cut or boosted. Low shelf filters, all frequencies lower than it will be affected. High shelf, all frequencies higher than the selected frequency will be affected.

Pass: Set the cut off frequency and past that point frequencies will be cut drastically.

Getting deeper into the shapes, let's start with Bell filters.

Proportional Q changes the Q factor in proportion to the amount of boost or cut that's applied. The less gain you have, the wider the Q or bandwidth. The more gain, the narrower the Q gets. This is similar to how an API equalizer functions.

Bell: The standard bell filter does not alter its Q factor while boosting or cutting.

Band Shelf is an interesting one. Instead of having a little peak, it has a flat top and then smooths out on the edges. If you want to boost a range of frequencies all at once, use this.

Shelves

Analog: Standard shelf in an analog EQ.
Baxandall: Gentle/transparent. A favorite of mine.
Vintage: Based on the Pultec design. Fatten low end without the mud. Adjust the Q.

Resonant: Emphasize the lowest and highest part of the signal. Adjust Q to make the fun/magic happen.

Pass Filters

Flat: Butterworth design. My favorite for high and low passing duties because it does so without changing the character of a signal.

Resonant: Boosts the signal at the cutoff frequency.

Brickwall: Elliptic design. Steep but minimal signal ripple. Use in vinyl mastering.

In Digital Mode, a Surgical shape option is added to all categories. This gives even more shaping options.

Hold shift to lock either frequency or gain adjustments, depending on which way you are dragging the node.

Advanced Mode: Enabled in Digital mode. 0% phase = linear phase. 100% = minimum phase.

To reset a band, double click the node.

Equalizer Settings Options

Alt Solo Filter Q is a setting to change to your preference!

Soft Saturation has a more analog clipping sound with EQ boosts.

Buffer Size: How many samples are stored in the memory buffer. May help with clicking/popping issues.

Frequency Resolution: The minimum that the EQ can be adjusted. Ask iZotope exactly what this means. The highest Q factor or something else? Maybe the resolution it adjusts by when changing the Q? (in manual, add digital mode only as the tooltip says)

Filter Size: The filter steepness/kernel length. Changes when resolution setting is modified.

Match EQ

is a revamped module re-release. So technically it isn't new but it's better than old.

iZotope recommends putting Match EQ as the first equalizer.

I recommend using the standalone module. That way, presets can be saved.

From the top, channel processing mode, three choices. Reset All button.

In the middle is our spectrum view, where the Action Region can be set with the cut off handles. Drag or double click the numbers By default, they are 20 Hz and 20 kHz. You can also drag the range in the middle.

Targeting an area may sound better than applying Match EQ to the whole frequency spectrum.

The idea is to load up a commercial mastered song in a separate project first, Hit play, capture it as a reference and then save the preset. If you didn't like what was captured, click Clear and try again.

Then, find a similar section to what you captured (chorus or verse) in the song you are masterng and when you hit stop it will adjust itself. Then adjust the Fine Tune controls.

Smoothing: Amount of precision. Lower smoothing is less precise.

Amount: How much processing is applied. Technically 100% is the closest match with 0% smoothing BUT it will sound unnatural.

Keep amount to under 50%. If the match curve has narrow peaks and valleys, use Smoothing to flatten them out.

The standalone module has some nice presets including Pink and brown noise AKA -6 dB Guide.

Maximizer (digital brickwall limiter)

My favorite Ozone module. If you have heard one of my mixes, chances are an Ozone Maximizer was used on it. It's a brickwall limiter, which makes it easy to bring the volume of mixes up without losing quality or going past a set level.

Choose from two viewing modes at the top left. Gain Reduction Trace or Spectrum Analyzer. The factory reset button is at the right.

On the left are the IRC modes. Each one has varying degrees of latency and CPU requirements. The lower down the list, the higher they are.

IRC LL is new for Ozone 9. It is similar to IRC I mode.

IRC I - The classic Maximizer mode. Good for its time but a relic now.

IRC II - Considering it was released in Ozone version 4, it still doesn't sound too bad if you don't push it past about 4 dB of gain reduction. Lighter on CPU than the newer modes. I'll use this when mixing.

IRC III - Ozone 5 introduced this mode. A good improvement upon IRC II in that distortion character style is selectable.

Pumping: Least aggressive.

Balanced: The mode I usually choose. As the manual says, constrains the release behavior in a transparent way.

Crisp: Aggressive contrains release behavior. Favoring distortion over pumping.

Clipping: Most aggressive style. Colors the mix a little and distorts to achieve loudness.

IRC IV: Introduced in Ozone 7. Builds on previous IRC modes, giving three options. Does so by limiting in specific frequency ranges, resulting in more transparency with greater perceived loudness.

Classic mode gives a sound that is similar to the older algorithms, particularly in the high frequency response.

Modern mode sounds more open/transparent. This is the one I usually go with.

Transient: If a musician demands a very loud master, this is the mode to go with in order to preserve dynamics such as the snare and kick drum.

The learn threshold control is nice for getting a ballpark loudness. The manual says do not rely on this for loudness standards compliance.

The True Peak option works for the most part BUT I think it rounds out transients a little, so use it with caution. Generally, I keep it turned off except for broadcasting compliance. Or if a musician demands a very loud peak for whatever reason.

Maximizer is very easy to use. You set your mode, for me is usually either IRC IV or IRC II. Then you set your ceiling, which is the maximum peak tat you want your audio to go to. Usually I have this set to -1.0 dB.

Then, hit play and find the loudest part of your song. Put it on a loop. Adjust the threshold down until you achieve the loudness that you want. with IRC II mode, you can get about 4 dB of gain reduction before i doesn’t sound transparent. With IRC IV, you can get away with more.

Character slider: I generally keep this on the default setting. It controls the overall attack and release times of the limiter. Lower numbers are faster, higher numbers are slower. Adjust to taste if necessary.

Stereo Independence:
I usually keep these controls on the default 0% and linked setting. If you want limiting to be applied differently to the left and right channels, this is the feature to use. New in Ozone 9 are the Transient and Sustain Controls. Previously, these were grouped together in one setting. Now, control how the limiter responds across the channels.

At 0%, the left and right channels are limited in exactly the same way. So, if something peaks on the left channel more, both channels are gain reduced by the same amount. At

100%, the right channel's peak limiting has zero effect on the left channel and vice-versa. Between 100 and 0, choose how much the channels differ.

A lop sided mix can occur at the 100% settings, which can sound unnatural. But, if you're going to use this setting it's ultimately a matter of taste. Tread cafefully though.

Transient Emphasis: How much transients are shaped before being limited. In practice, if your drums aren't popping like they were before limiting, try this feature. Dynamic perception will be preserved.

Dither

What is dither? Essentially, it's noise that hides digital distortion when bit depth truncation must occur. Inside of a modern DAW, audio is handled in floating point quality until it reaches the audio interface's outputs.
At 24-bit, this isn't an issue at all. The option for 24-bit dither is still there though. At 16-bit resolution, which has a -96 dBFS digital noise floor, it might be an issue.

So, dither is added to improve the audio quality just a little bit. Noise to mask distortion. It certainly sounds better in my opinion than digital distortion. If for some reason you have to encode at 8 or 12 bit audio, dither is essential.

The top section of the module shows the general noise shape of the dither. Once upon a time Ozone had different dither types to choose from, but now they only have MBit+ . Which is fine, because it is superior to the others. It's the best if not top 3 dither out there.

This module also includes the DC offset filter.

On the left, choose your destination Bit Depth. Typically this will be either 24-bit or 16-bit. Then pick the amount, how much noise shaping

Auto Blanking mutes dither after 700 ms of digital silence.

Amount: Medium is recommended. Strong raises the noise floor a little but completely eliminates any possible digital quantization ditortion.

You can really hear the differences if you switch it to 8 bit, auto blanking off at music fade outs.

If you don't choose any dither amount, the Harmonic Suppresson option is available. It moves the distortion away from overtones in audible frequencies.

Do the random nature of dither, some noise can peak louder when shaped. If you have it set to the stronger settings, consider enabling the Limit Peaks option.

Noise Shaping effectively moves dither noise to less audible frequencies. At high sample rates, it will be virtually inaudible.

My personal favorite settings are Medium Amount, High Noise Shape, Auto Blanking off and Limit Peaks on.

The bit meter is cool to look at. The inner columns display real time bit activity. The outer columns show peak hold information, duplicating the inner columns. That's why there is a reset button.

To use the DC offset filter, the Maximizer or Vintage Limiter module mustbe in use. A 1 Hz cut off filter is added when it's enabled.

The offset can happen when DC current leaks into the audio. It will offset it from the normal 0 voltage average. At worst, clicks/pops can be heard at the beginning or end of an edit. At beast, it adds inaudible noise floor to your track. I usually keep this off unless there are audible pops when starting mix playback.

Reference - Advanced Only feature

Allows for easy comparison between the audio you are working with and other tracks. Up to 10 reference tracks can be brought into the app or plugin.

Supported file formats are wav and aiff of course. But also FLAC, MP3 and AAC.

Tabs pop open and you can add more, re-order them or remove.

When a song is loaded, loop segments are automatically generated. These allow you to choose certain areas of a track for easy reference. Say, a chorus or verse.

Region Selector handles can be moved left or right. Zoom in and out using the mousewheel for precise placement.

You can name the loop segment by clicking the name. By default, they are A B C D, etc.

To add a loop, simply right-click on the waveform area where you want the new loop to begin and select Insert Loop.

To delete a loop, right click on the one you want to remove and select Remove Loop.

To A/B your track, click the Power button next to the Reference panel button. (lower right)

Gain is another important control. when A/Bing, you will usually want the volume levels to match. Use the slider, hold Control for precision adjustment or type in the gain. Double click or alt-click resets to zero. Gain is set on a per track tab basis.

To make this a little easier, click the I/O button above the main meters. Change Type to Momentary or Integrated. Take note of your track's loudness levels. Then click I/O again and enable the Replace Input with reference option. Set gain based on this.

To reference the whole track, click Select All at the bottom right.

Low End Focus

A new module in Ozone 9. Does what the name implies...focuses on the low end AKA the bass section of a mix. If there are issues, this may help fix them.

As the manual says, it's designed to reduce mud, increase impact and address other common low end issues.

Only one view with this module. So, in the Header area we have the standard channel processing mode selector along with the Reset button.

Select the range of frequencies that you want processed. Use the S solo button to aide in the search. The Solo output is before processing. iZotope calls this the Action Region.

Adjust by dragging the handles, dragging the region itself or by double clicking the numbers below the handles and entering a new number between 20 and 300.

Down in the Controls section, there are just a few but you need to memorize what they do.

Modes: Punchy is the more obviously audible mode. Whether you use positive or negative contrast, they are affected similarly. Smooth mode allows for a subtle alteration of the low end.

Positive contrast: Brings out transients by reducing spectral content that lack isn't apart of the transient. So, if you can't hear the kick drum easily this is the setting to choose.

Negative: De-emphasizes transients by blurring the low end. Similar to analog compression saturation. If you hear too much kick drum, choose this option.

Listen: Allows you to hear what changed between the unprocessed and processed signals.

Gain: Makeup gain. When using negative contrast this especially important.

Master Rebalance

In the TV industry, the saying goes “Give the people what they want.” Ozone 9 tutorials are apparently not what you guys want, so I’m going to wrap the video series up short. The rest of my Ozone 9 guide will be made available in text form though.

This is the new miracle worker module in Ozone 9 Advanced. It allows you to change the vocal, drums and bass levels of a stereo mixdown. Similar to what can be accomplished during stem mastering.

It’s very easy to use. Pick which element of the mix you want to adjust, then boost or cut its volume with the Gain slider. That’s it. Hold control while dragging for precise adjustment.

To reset controls, upper right loopy button.

The blue spectrum is the focus instrument and the gray spectrum is the leftover signal.

In my opinion, it does an excellent job on bass and drums. It does an OK job on vocals. If you need it for about 3 or 4 dB then it’s fine...just don’t expect results as good as the drums and bass detection. It will give you a good indication of what to tell the mix engineer though.