HOME STUDIO MASTERING

The One Hour
MASTERING WORKSHOP

Free plugin bundle included
VST / VST3 / AU / AAX

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*A 5-step mastering method*
for home studio producers

Index

Introduction

In June 2018 my book Home Studio Mastering was published by publishing house Routledge / Focal Press. Home Studio Mastering is a step-by-step manual that gives you all the tools to professionally master your music yourself. It demystifies the subject in a hands-on way for those working in a home studio and provides comprehensive guidance, from buying equipment and applying acoustical treatment, to using different audio applications and mastering plug-ins.

The book is divided into three parts. The first part of the book describes the criteria a mastering studio must meet and provides ways to set up your home studio accordingly. The second part provides an analysis of all possible mastering techniques for optimizing the tonal balance, width and depth, dynamics, and loudness of your music. The third part of the book is a mastering workshop called *The One Hour Mastering Workshop* and guides you through the whole mastering process step-by-step in about one hour, using the free bundle of five mastering plug-ins that was specifically developed to accompany the book: the *Noiz-Lab LE Mastering Bundle*.

This eBook contains the full text of the *One Hour Mastering Workshop* from the book, and can be seen as comprehensive and workflow-based manual to the *Noiz-Lab LE Mastering Bundle*. It will guide you through the whole mastering process in about one hour, using only these five mastering plug-ins.

For more info about the full book and accompanying mastering bundle, go to www.homestudiomastering.com.
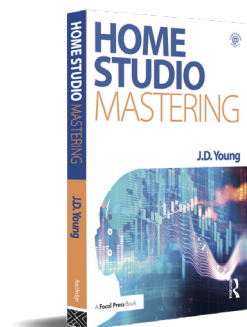
Download the mastering bundle

Before you can start the One Hour Mastering Workshop, you need to install the Noiz-Lab LE Mastering Bundle. You can download the plugin bundle for free from my book's website: www.homestudiomastering.com. Depending on your system, download either the PC or Mac version and unpack the zip-file. Then, make sure your sequencer or audio editor is closed, and run the installer. You will be presented with a standard installation dialog that will automatically install the five Noiz-Lab plug-ins in the appropriate folders on your system. Although the installer allows you to choose custom VST folders on PC, and allows you to deselect certain plug-in formats, it is best to just stick to the default installation, unless you have specific reasons not to do so. If the installation was successful the plug-ins will be added to the plug-in menu in your sequencer or audio editor, either listed alphabetically or in a developer subfolder called 'Noiz-Lab'. If the installation was unsuccessful somehow, or the plug-ins do not show up in your sequencer or audio editor, read the FAQs on the website, or use the website's bug report form or contact form to get support.

The plug-ins and 5-step workflow

Noiz-Lab LE Mastering Bundle

In order to quickly and easily create a professional master, the five plug-ins in the *Noiz-Lab LE Mastering Bundle* are designed to work together in a very specific way. Before starting your mastering session therefore, it is important to understand the role each plug-in plays in the *Noiz-Lab LE 5-step Workflow*:

1 Monomizer

The *Noiz-Lab LE Monomizer* is a stereo imaging plug-in that lets you narrow the low end stereo image of your track. It gives you full control over the low end frequency range you want to affect, as well as over the curve and amount of the stereo-to-mono effect. In addition to its stereo imaging functionality, the Monomizer plug-in also features a

filter that will reduce DC offset problems dynamically over time. This plug-in is first in the chain, because its stabilizing effect on the stereo image, and possibly on the tonal balance as well, will make the remaining processing more effective. Removing DC offset problems is also done at the start of the mastering chain, because the asymmetry introduced by more extreme DC offset can negatively affect the workings of dynamics plug-ins that make use of a threshold, like the Compressor and Limiter further up in the chain.

2 Equalizer

The *Noiz-Lab LE Equalizer* is an analog style minimal phase equalizer, for optimizing your track's tonal balance and stereo image. The plug-in features six filters that can be individually set to have a low pass, high pass, low shelf, high shelf or peak response, and to process in stereo, mid or side mode. It is possible to switch the filter type and processing mode of all filters individually, allowing you to quickly and easily try different equalization approaches. In addition, the Equalizer features a real-time spectrum analyzer for visual feedback, and lets you listen back to just the mid or side channel of your track, which allows you to analyze and adjust the tonal balance and stereo image of your track in more detail. The Equalizer plug-in is second in the chain to ensure that the tonal balance of your track is optimized before using any dynamics plug-ins, like the Compressor and Limiter further up in the chain, that might otherwise react to unwanted frequencies.

3 Compressor

The *Noiz-Lab LE Compressor* is a dynamics plug-in that lets you apply four commonly used mastering techniques for controlling dynamics and adding character: compression for glue, balance, punch and impact. As these compression styles depend on a large number of specific parameters and specific values, most functionality is taken care of behind the scenes. This makes it possible to quickly and intuitively compare compression styles, using just the threshold control to vary intensity, and the gain reduction meter for visual feedback. In addition, the compressor features a saturation control to add character to your track, and a make-up gain control for doing an easy before-and-after comparison on the same perceived loudness. The Compressor plug-in is third in the chain, after the Monomizer and Equalizer, to ensure it won't react to unwanted frequencies, and before the Convolver and Limiter, that will both react more naturally to a track with more controlled dynamics.

4 Convolver

The *Noiz-Lab LE Convolver* is a convolution plug-in that lets you impose the tonal balance and acoustic properties of different real-life rooms onto your track. You can choose between a studio, hall and church response, that will all create a certain depth and add a specific timbre or character to your track. You can mix the convolved signal in with the amount control, and use the make-up gain control for doing an easy before-and-after comparison on the same perceived loudness. In addition, you can shape the tonal balance of the wet convoluted signal with a high pass and low pass filter, and choose between a mono and stereo impulse response. The Convolver plug-in is fourth in the chain, after optimizing the tonal balance and dynamics, to ensure the convolution process produces a balanced and subtle result, and before the Limiter, which needs to be the last plug-in in the chain.

5 Limiter

The *Noiz-Lab LE Limiter* is a brickwall limiting plug-in that lets you maximize the loudness of your track. Although most of the trickery involved in reducing limiting artifacts - like distortion and breathing-and-pumping - is kept behind the scenes, you can choose between a fast, medium and slow auto-release, and between different modes to boost the transients in your track. With the maximize control turned on, turning down the limiter's threshold will maximize your track's volume up against the ceiling value. However, with the maximize control turned off, the volume will not be automatically boosted. Preventing loudness bias like this, will make it easier to hear any degradation caused by the plug-in. The Limiter plug-in is fifth and last in the chain, after all other optimizations have been done, to minimize artifacts and degradation, and to ensure your track will not clip on the master bus.

Prepare your mastering session

Once you have familiarized yourself with the Noiz-Lab LE plug-ins and workflow, it is time to render your track and set up your mastering session. If you want to keep to the default recommendations of the full book Home Studio Mastering,, remove all plug-is on the master bus of your mix project, make sure your track doesn't clip, and render your track as a stereo .WAV or .AIFF file on a samplerate of 44.1kHz and a bitdepth of 24bit. Then, open a new project in your sequencer or audio editor for your mastering session, and set this project to 44.1kHz and 24bit as well. Import your track and reference tracks on different channels, stacked vertically for doing instant comparisons using the solo option. If your reference tracks have a different samplerate and bitdepth than the project, it is fine to let your sequencer or audio editor convert them to the project settings. Create a group channel and route your reference tracks to it, so you can control their volume with just one fader. As they are probably already mastered, lower their volume a little to create some headroom, say by about 12dB. Then, use the reference tracks' individual faders to match their loudness to each other by ear. And finally, use the reference tracks' group channel fader to match their volume to the track you are about to master. Although your mastering session is now set up, after repeatedly adjusting the volumes of your reference tracks, the chances are you have lost your objectivity. Therefore, before you actually start your mastering session, it is a good idea to first take a short break, and then re-calibrate your brain by listening to your reference tracks for about 10 to 15 minutes at a comfortable volume. After doing that, let's get started.

Step 1
Monomizer

Goals

Removing DC offset dynamically over time

Tightening up the low end stereo image

Load up the Monomizer plug-in on your track's channel, making sure it is the first plug-in in the mastering chain. You are going to use the Monomizer plug-in to remove potential DC offset, and to tighten up the low end stereo image of your track. Removing DC offset is usually not necessary, except when the offset is more extreme, say a few percent or more. Most sequencers and audio editors have an analysis tool that shows you the DC offset of your track. If the analysis shows a significant offset, you can simply activate the Monomizer's DC offset filter to compensate for this. Tightening up the low end of your track, on the other hand, can be more of a challenge, because the low end generally poses the most issues for home studio users. For the purpose of this first step therefore - and for doing any low end processing during this mastering session - keep the following recommendations in mind:

► Find out the range or frequency response of your speakers, so you are aware of any limitations that could cause trouble. If you followed the default recommendations of the full book, you are using decent near-field speakers with 6 to 8 inch woofers. Such speakers will usually go down to about 40Hz, which is certainly manageable if you use the remaining workarounds in this list as well.

► Double-check all low end processing you do on a professional pair of headphones which has a frequency response down to 20Hz (the low limit of the human audible frequency spectrum). Using a pair of headphones like this, might not give you a highly accurate response in the sub-low end, but it will clearly reveal issues otherwise obscured by acoustic problems in your room or by the limitations of your speakers.

► When working on the low end, turn up the volume. Listening at a higher volume will exaggerate the low end (and high end) in a psycho-acoustic manner, which makes it easier to analyze the low and sub-low frequency range. If you are only concerned with the low end, you can insert an instance of the Equalizer plug-in after the Monomizer and filter out the high frequencies with a low pass filer. This will make analyzing the low end much easier, especially in the sub-low range.

Before choosing your settings for the Monomizer, it is a good idea to compare the low end stereo image of your track to that of your reference tracks, so you have a clear reference. Primarily use your headphones for this with the volume turned up, but double-check your findings on your speakers as well. For a more detailed analysis, experiment with a temporary low pass filter to isolate the low end for playback. In general, commercial tracks featuring drums and bass will be mostly mono in the low end from about 100Hz, and fully mono approaching the low limit of the human audible frequency range of 20Hz. That is, when no stereo effect is used to deliberately widen the bass or bass-synth parts. On the other hand, commercial tracks that do not feature drums and bass, or put less emphasis on the low end, will usually roll off the low end, to clean up the track. If this is the case, it still is a good idea to analyze the low range of any lead instruments like guitar, piano or synth, as their fundamental frequencies might be low enough to make it worthwhile to slightly narrow their low end stereo image. Go through the following steps:

► To get a feel for the Monomizer effect, turn up the volume of your track, set the low frequency control to 20Hz and the high frequency control to 320Hz, and turn the make mono knob slowly to 100%. You will notice that any mix elements in the low end, gradually move closer to the center of the stereo image, due to a summing based stereo imaging algorithm. Skip through the different shapes to hear how they affect the stereo-to-mono transition along the frequency spectrum. The differences between shapes are subtle, and therefore best detected by using headphones and a low pass filter to isolate the low end.

► Now, determine if your track actually benefits from narrowing the low end stereo image. If the low end already is predominantly mono, this might not be the case. However, if the low end has a lot of stereo information, or the fundamental frequency ranges of any lead instruments spill down to the low end, your track might benefit from a tighter low end stereo image, as this will stabilize your track's overall stereo image and possibly the tonal balance as well. In that case, proceed with the next two steps.
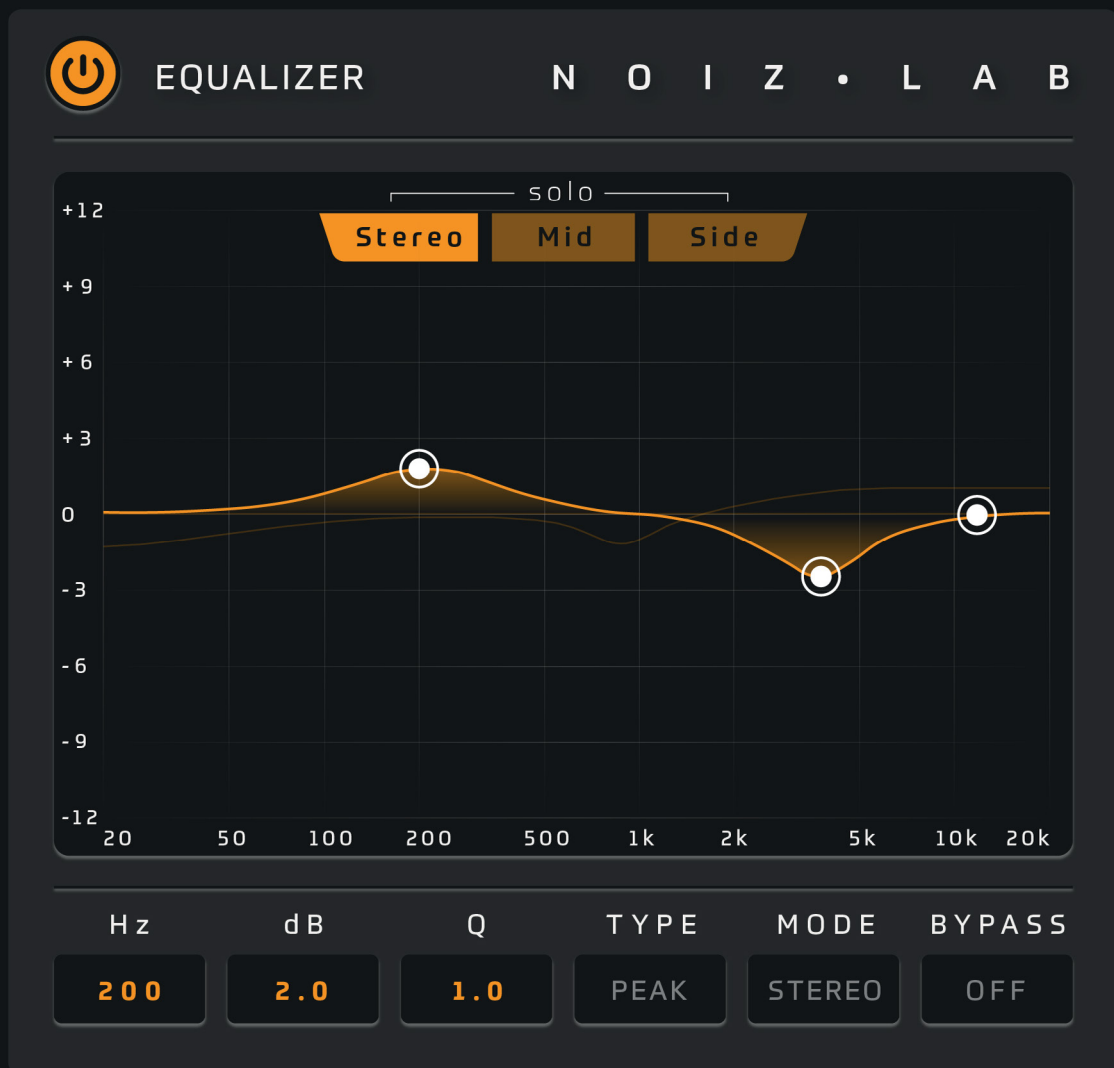
► If you already like your track's low end stereo image compared to your reference tracks, or you have trouble determining the best course of action, it is best to take a conservative approach. Set the low frequency knob to 20Hz, the high frequency control to 100H and the shape to S-curve. Then slowly increase the make mono control up to a maximum of about 50%, for a subtle stereo-to-mono effect. As frequencies below 100Hz are generally perceived as mono anyway, this will add a little more stability to your file, without the risk of harming your track.

► If the low end of your track sounds a bit messy compared to the reference tracks though, raise the low frequency to 40Hz, raise the high frequency to about 160Hz and raise the make mono control to a maximum of about 75%. In addition, experiment with the different shape settings to find the best stereo-to-mono transition along the frequency spectrum. Be careful not to take it too far though, as this might cause your track to lose power and punch. For more extreme settings like this, make sure you carefully compare your track to your reference tracks.

As mentioned at the start of this step, the low end generally poses the most issues for home studio users. However, if you are uncertain about your Monomizer settings at this point, don't worry too much. We will revisit the low end several times during the remaining steps of this mastering course. In addition, the remaining steps will require you to listen to your track in different ways, gradually revealing the best general approach to mastering your track. In that regard, the chances are that you will go back and forth a few times between plug-ins before reaching the end of this mastering course.

Step 2
Equalizer

EQUALIZER N O I Z • L A B

solo

| Stereo | Mid | Side |

+12
+9
+6
+3
0
-3
-6
-9
-12

20 50 100 200 500 1k 2k 5k 10k 20k

Hz	dB	Q	TYPE	MODE	BYPASS
200	2.0	1.0	PEAK	STEREO	OFF

Goals

Optimizing the tonal balance

Improving the stereo image

Load up the Equalizer plug-in on your track's channel, making sure it is inserted in the mastering chain after the Monomizer. Because the Equalizer plug-in can process in stereo, mid and side mode, you are going to use it for both optimizing the tonal balance of your track, as well as improving its stereo image in a very natural way. The biggest challenge with making these optimizations in a home studio, like explained in the previous paragraph, is controlling the low end. Particularly when it comes to sub-heavy genres like Hip-hop and EDM, you need to take special care, because it is usually the well-balanced sub-bass that gives these genres their powerful sound. As controlling the low end will make all other processing techniques to come more predictable, and therefore easier to perform, that's where you will start. Before touching any of the plug-in controls though, take some time to compare the low end of your track to that of your reference tracks again. As with the Monomizer, primarily use your headphones for this with the volume turned up, but double-check your findings on your speakers as well. Then, go through the following steps:

► Select the outer-left filter, set the filter type to high pass and the processing mode to stereo. If you move the frequency up and down the spectrum, either by dragging the node or the frequency control itself, you will gradually hear the low, then the mid and finally most of the high frequency content in your track disappear. Using your headphones, set the frequency to 20Hz, and listen carefully how the low end timbre changes, when you very slowly raise the frequency to a maximum of about 100Hz, and see if you can match the timbre of your reference tracks.

► If you are somewhat experienced, and have a nicely treated room and a good set of speakers or headphones, I recommend being conservative at this point, especially with sub-heavy genres. Set the frequency between about 20Hz and 30Hz, but make sure you are not losing any power due to the roll-off of the filter. This will clean up the low end slightly, and increase the potential maximum loudness of your track by removing any sub-low frequencies that may have very high amplitudes.

► If you are not that experienced and don't know what to listen for exactly, I recommend being a little more aggressive to be on the safe side. Set the frequency between about 30Hz and 40Hz, to make sure you won't encounter any problems when playing your track on bigger systems. For tracks without (kick-)drums and bass, like a singer-songwriter song or piano ballad, you probably want to go a little higher, up to about 50Hz. Make sure though, you are not losing any warmth. Don't worry about any remaining uncontrolled low end in comparison to the reference tracks at this point, as you are going to have a few more opportunities to compensate for this.

Next, to ensure your track will translate well to the broadest possible variety of audio systems - from tiny mobile phone speakers to huge PA systems - you will need to make sure your track's overall tonal balance matches your reference tracks and has no significant inaccuracies. First though, take some time again to compare the tonal balance of your track to that of your reference tracks, primarily using your speakers. Do make sure though, that your reference tracks are accurately level matched to your own track, to avoid any bias caused by the psycho-acoustic relationship between loudness and tonal balance. Start by comparing the tracks in regular stereo mode, to determine if the overall tonal balance differs from the reference tracks in any significant way, and if actual corrections and improvements to the overall tonal balance are needed. Then, use the solo button on top of the Equalizer plug-in to listen the isolated mid and side channel of your track. To compare your track's mid and side channels to that of your reference tracks, insert a temporary instance of the Equalizer plug-in on the reference channel's group channel, to switch between playback modes for all reference tracks at once. Comparing the mid and side channels will give you a good indication of how the stereo field of the reference tracks transitions along the frequency spectrum. Just as with the overall tonal balance, try to determine how your track's mid and side channels differ from that of your reference tracks, and if there are opportunities to enhance the stereo image. To create an initial balance, go through the following steps and make the necessary adjustments:

► Compare your track's overall tonal balance to that of the reference tracks in regular stereo mode, and listen to the low and high end. If your track sounds overly dull or bright in comparison, use a stereo low and high shelve to create a general balance. Make sure the curves of the shelves are smooth to avoid any resonant characteristics. If your mix is decent, you shouldn't need to boost or cut the stereo shelves more than about 1dB to 2dB.

► Listen to the mid channel of your track, and use the high and low shelf filters to make sure there is enough low end present, and no excess of high end present. In addition, listen to the side channel of your track, and make sure the opposite is true: enough high end, and no excess of low end. If your mix is decent, you shouldn't need to boost or cut the mid or side shelves more than about 2.0dB to 3.0dB. Apart from improving the tonal balance, the combination of these techniques can improve your track's stereo image in a very natural way.

► Carefully level match and compare your track to the reference tracks again, both on your speakers as well as on your headphones, to make sure your settings have the intended effect. Although you might need to revisit your settings later on, this initial general balance will provide you with a solid foundation for making any other adjustments to the tonal balance.

Now a general tonal balance has been established, determine if your track contains any unwanted or resonant frequencies, or lacks certain important qualities. For example, your track might sound overly muddy or boxy, or nasal or harsh in comparison to your reference tracks, or may lack power, warmth, fullness or definition. For every problem you define, go through the steps below. In case you have used most of the Equalizer's filters already, it is best to insert an additional instance of the plug-in, to keep the different equalization phases discussed in this paragraph separate and accessible.

► Find out if the problem is in the stereo, mid or side channel of you track. If the problem is in the mid or side channel, solo the channel so you can more accurately hear what you are doing. Select one of the filters and set its filter type to peak. In addition, make sure it's processing mode is set to the channel you want to affect. Then, create a boost by raising the node of the peak filter to the top of the window. Move the node slowly to the left and right until you find the target frequency.

► Once you have identified the target frequency, set the Q-value with the goal to isolate the problem area without affecting the rest of the channel. As you want your filter to have a smooth curve to avoid any resonant qualities, a Q-value of 1.0 or less is generally considered appropriate for mastering. If you have a very specific problem though, you can try going up to a maximum of about 3.0 for a more surgical approach.
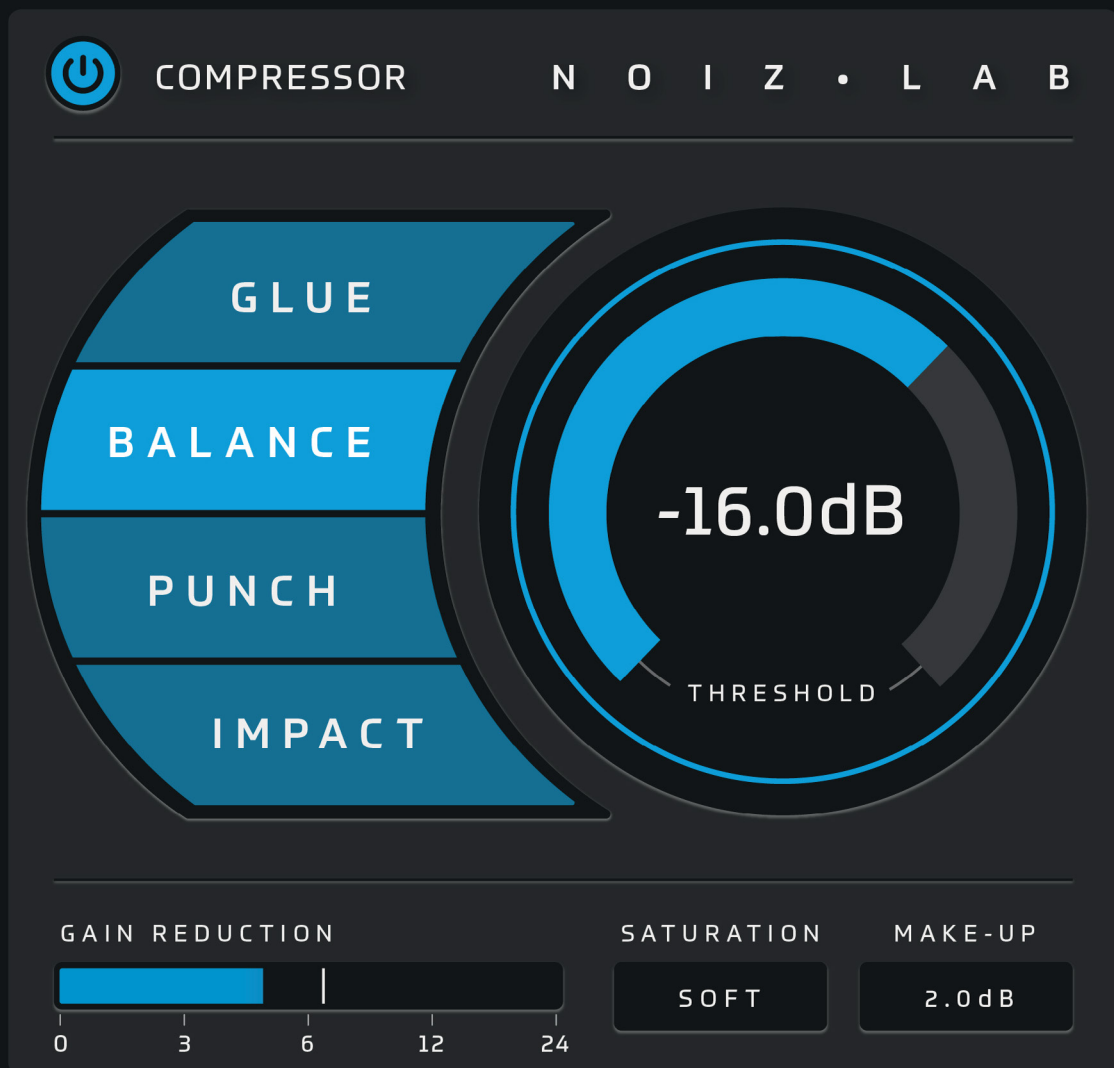
► After setting the Q-value, decide on how much you want to boost or cut the target frequency. After boosting the filter to find the frequency though, it is easy to fool yourself. Therefore, take a moment to listen to your reference tracks again, before applying the final gain setting. To avoid moving the handle sideways, and losing your focus on the frequency, it is easiest to drag down the dB edit box control instead of the node itself. If your mix is decent, you shouldn't have to boost or cut more than a few dB. Actually, peak filter adjustment of only about 0.5dB to 1.0dB are probably the most common in mastering.

After matching your reference track's tonal balance, you might feel you are doing harm to your track, when you do a before-and-after comparison. If this is the case, think carefully about the goal of the mastering session. Your goal is to create a master that translates to a wide variety of audio systems, which requires a tonal balance without significant inaccuracies that holds to the conventions of its genre. In that regard, doing a before-and-after comparison is often irrelevant, as it tells you nothing about the objective balance of your track. A before-and-after comparison only tells you something about the relative balance of your track compared to an earlier version, which makes your assessment of the effectiveness of your settings extremely vulnerable to psycho-acoustic bias. Therefore, during this mastering workshop, try to mainly focus on the objective balance of your track in relation to your reference tracks,

but without compromising on your artistic vision. In addition, keep in mind that this is only the second step of our mastering session, and that you are comparing your track to fully-mastered reference tracks. Therefore, don't worry if your track doesn't match up to your reference tracks just yet.

Step 3
Compressor

COMPRESSOR   N O I Z · L A B

GLUE

BALANCE

PUNCH

IMPACT

-16.0dB

THRESHOLD

GAIN REDUCTION

0   3   6   12   24

SATURATION

SOFT

MAKE-UP

2.0dB

Goals

Controlling dynamics

Adding character

Load up the Compressor plug-in on your track's channel, making sure it is inserted in the mastering chain after the Equalizer or multiple Equalizer plug-ins you have used during step 2. You are going to use the Compressor plug-in to control your track's dynamics, and to add character by subtly applying the Compressor's built-in saturation feature. The Compressor contains four compression styles, that are discussed below. Try them all, and determine if they benefit your track. Keep in mind that both the compression and saturation effect will influence the volume of your track, and in many cases the perceived loudness as well. Make sure therefore, to use the make-up gain on the plug-in to compensate for any volume loss. This will enable you to do an objective before-and-after comparison to see if your settings don't cause distracting volume changes or have a negative effect on your track's transients, as well as objectively compare your track to your reference tracks.

Compression for glue

This technique will help to glue your file together and add some subtle fullness and warmth. It is a very basic technique, but it can add a nice bit of stability to your file. If you hear someone talking about general mastering compression, this technique is probably close to what they mean. Set the mode to glue. Play the loudest part of your file and pull down the threshold until you get a maximum of about 2.0dB to 3.0dB gain reduction. Experiment with the different saturation settings to soften the transients and add a little warmth to your track. Adjust the make-up gain and do a careful before-and-after comparison.

Compression for balance

This is a parallel compression technique that will bring out the softer parts of your file, which can be very helpful if you want to subtly reduce the dynamics of your file without making it sound unnatural. It is especially useful for guitar music and other acoustic genres, like singer-songwriter and classical music. Set the mode to balance. Play the loudest part of your track and pull down the threshold until you get a maximum of about 5.0dB gain reduction. Experiment with the different saturation settings to even out the dynamics a little more. Adjust the make-up gain and do a careful before-and-after comparison.

Compression for punch

This is a subtle technique to add some more punch to your track by giving the transients - like the kick and snare - a little boost. It works very well for pop and rock music with a drum-heavy mix. Set the mode to punch. Play the loudest part of your track and pull down the threshold until you get a maximum of about 1.0dB to 2.0dB gain reduction. Everything except the transients will be brought down a little. Experiment with the different saturation settings to add a little grit to the signal. Adjust the make-up gain and do a careful before-and-after comparison.
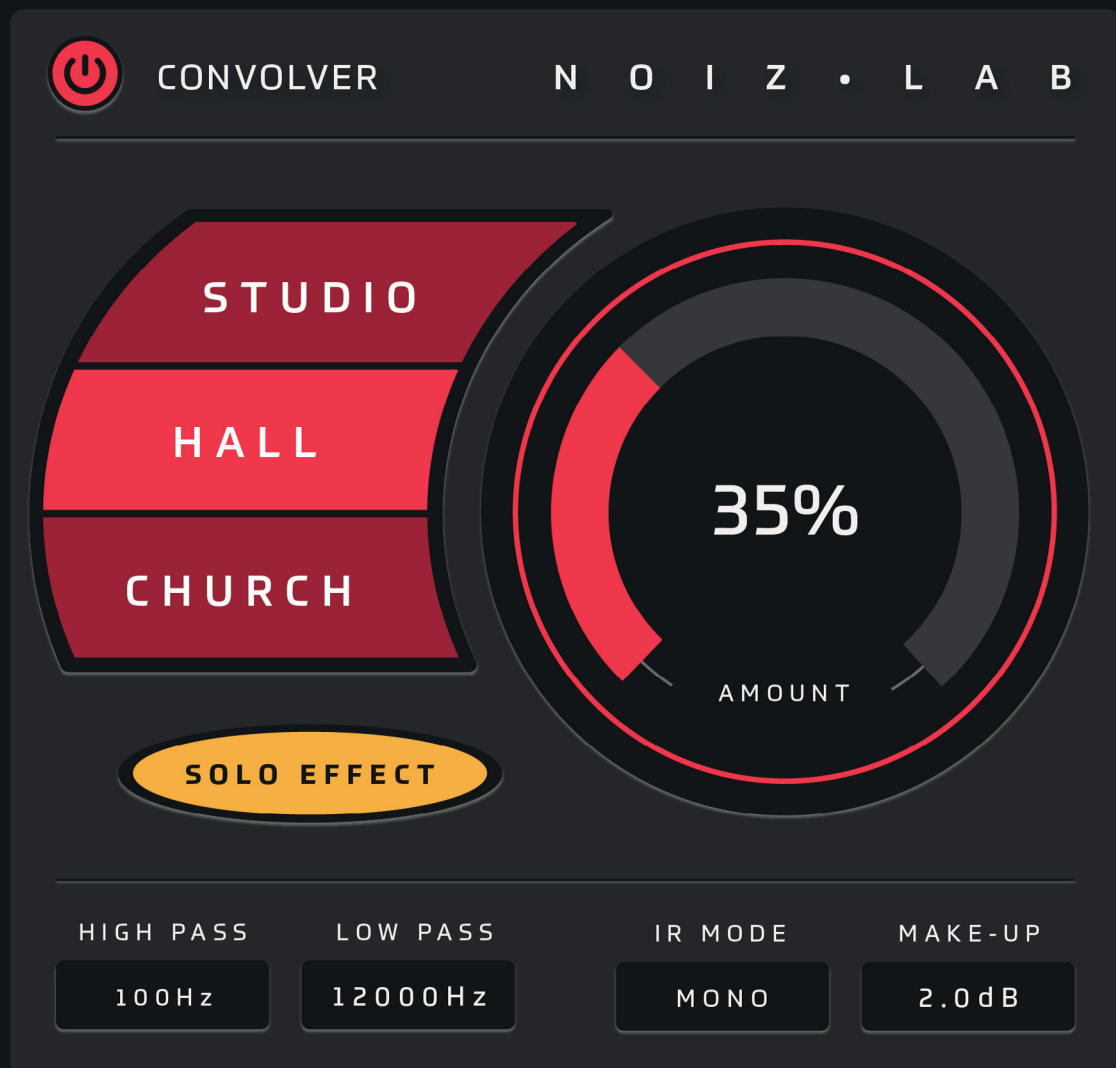
Compression for impact

This is a parallel compression technique based on the New York compression principle. It is far more vigorous than the previous compression techniques and is perfectly suited to give loud and powerful music more impact and a modern and produced sound. Set mode to impact. Play the loudest part of your track and pull down the threshold until you get an overly compressed sound where the transients begin to sound explosive. Experiment with the different saturation settings to even out the dynamics a little more. Adjust the make-up gain and do a careful before-and-after comparison.

After performing objective before-and-after comparisons to make sure your compression settings don't do any harm to your track, as well as objective comparisons with your reference tracks to make sure you hold to the particular genre's conventions, you should have a clear idea if any of these four techniques is beneficial to your track. If more than one technique gives a positive result, load multiple instances of the Compressor in your chain and experiment with the order of compression techniques. Combining compression techniques like that is common practice, because applying several subtle techniques consecutively usually sounds more natural and controlled than applying one technique in a more extreme way. In addition, inserting an Equalizer instance after a Compressor to subtly compensate for the compressors coloration, can help in achieving a controlled and balanced master. In general, for more powerful music genres, like rock, Hip-hop and EDM, you probably want to predominantly use the compression for impact or compression for punch technique, or a combination of both. On the other hand, for more subtle music genres, like singer-songwriter, folk and more classical oriented styles, you probably want to predominantly use the compression for glue or compression for balance technique, or a subtle combination of both.

As you have noticed, so far you have just been making small improvements with every step, which is typical for a standard mastering session. There are several reasons for this. Firstly, because you want your processing to be as transparent as possible, meaning that you want to get the intended results without the plug-ins themselves being noticeable. Secondly, as discussed above in the context of compression, several subtle adjustments will almost always sound more balanced than one big adjustment. And finally, applying all these different techniques and listening in different ways will make it gradually more clear what the best approach is for mastering your track. After completing this step, your track should start to sound relatively balanced, both in terms of tonal balance as well as dynamics. This will make the result of the remaining two steps predictable and stable.

Step 4
Convolver

CONVOLVER N O I Z • L A B

STUDIO

HALL

CHURCH

SOLO EFFECT

35%

AMOUNT

HIGH PASS	LOW PASS	IR MODE	MAKE-UP
100Hz	12000Hz	MONO	2.0dB

Goals

Creating depth

Adding character

Load up the Convolver plug-in on your track's channel, making sure it is inserted in the mastering chain after the Compressor or multiple Compressor plug-ins you have used during step 3. You are going to use the Convolver plug-in to create depth and add character to your track by convolving different impulse responses (IRs) of real world spaces with your track. The Convolver contains three different IRs, of respectively a recording studio, a concert hall and a church. All three IRs are deliberately shortened to be suitable for mastering, and can be used in mono as well as stereo mode. In most cases you want to use the mono IR version, to preserve your track's stereo image. However, if your track is mostly mono itself, using the stereo IR version, can create a very natural stereo image in a way that is difficult to achieve with any other processing technique. Go through the following steps:

► To get a feel for the Convolver effect, set the mode to studio, set the high pass frequency to 20Hz and the low pass frequency to 20,000Hz. Then, press the solo effect option and turn up the amount to 100%. You are now hearing the fully wet convolved signal, as if your track was played in the actual room the IR was taken in. Skip through the studio, hall and church mode, and switch between the mono and stereo versions of the IR to get a good feel for the different characteristics they provide, both in terms of tonal balance and reverberation. Use the filters to shape the tonal balance of the wet convolved signal, allowing you to blend in the wet signal with your track more seamlessly.

► Now, determine if your track actually benefits from using the Convolver. Because the convolution process imposes the acoustic properties of an actual space onto your track, it is ideal for enhancing the overall acoustic timbre of tracks that are purely produced in-the-box or recorded exclusively with close-miking techniques. Any of the three modes might provide a suitable character, and add the appropriate depth to your track. For tracks that already have an outspoken character or sound appropriately spacious, you might want to try the studio mode, as it is a subtle IR that can add a nice subtle flavor or help with creating a loud master.

► If you feel one of the IRs benefits your track, solo the wet convolved signal and use the filters to shape the wet signal's tonal balance. In most cases when using raw IRs, you want to take out some of the low end to avoid possible low rumble. Removing up to 100Hz should solve the low rumble, but if you are aiming for a specific timbre, you might want to raise the cutoff frequency to taste. The same goes for the high end. If you want the convolved signal to blend in nicely with you track, you probably want to take out some of the high end to make the IR less bright, or lower the cutoff frequency to taste.

► If you have shaped the wet convolved signal to your liking, set the amount to 0% and un-solo the effect. Then, slowly mix in the wet convolved signal with your track. Because adding the convolved signal will raise the overall volume, make sure you compensate for this with the make-up gain. In addition, try both the mono and stereo version of the IR. In most cases the mono version will give the most stable results, but if your track has a narrow stereo image, the stereo version might give it a more spacious sound, in a very natural way.

Because adding the wet convolved signal has changed the tonal balance of your track, you might want to compensate for this with a new instance of the Equalizer. Usually, applying a subtle low or high shelf filter, or both, should suffice. Only if the IR introduces some kind of specific coloration or resonance frequency, making a subtle cut with a peak filter might be appropriate. Make sure you keep primarily comparing your track to your reference tracks, instead of doing a before-and-after comparison with just your own track. In addition, in preparation of maximizing the loudness of your track during step 5, you can use the Equalizer to boost certain frequencies that are important for psycho-acoustic loudness as well. Be careful though, not to disturb your track's tonal balance too much. Go through the following steps:

►Select a peak filter and make a boost at 1,800Hz of about 1.0dB to 1.5dB, with a Q-value of about 0.5. This frequency is often boosted in rock and metal productions, for a loud and edgy sound. Make sure though, that your track doesn't start to sound overly thin and sharp.

►Select a peak filter and make a boost at 5,000Hz of about 0.5dB to 1.0dB with, with a Q-value of about 1.0. Humans are very sensitive to this frequency range, because this is where the intelligibility region of speech is found. Make sure though, that your file doesn't start to sound overly bright.

►If the low end is not very important to your track, select a low shelving filter and make a cut at 250Hz of about 0.5dB to 2.0dB with a very smooth curve. While this doesn't actually do anything at this stage, it will give the Limiter in step 5 a little more headroom, as low frequencies have high amplitudes and usually reach the threshold of compressor and limiter plug-ins first.

Step 5
Limiter
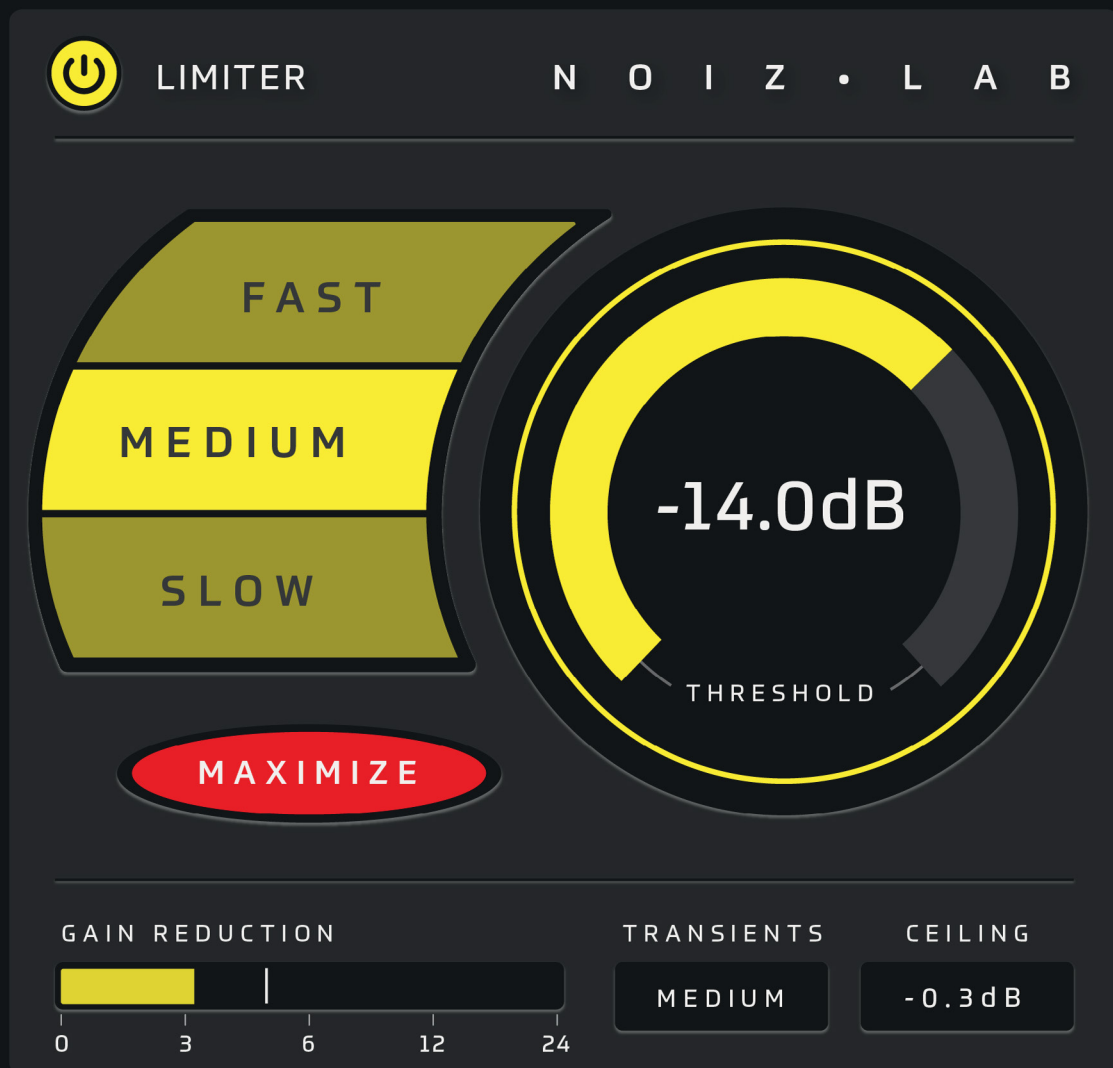
Goals

Maximizing loudness

Preserving transients

Load up the Limiter plug-in on your track's channel, making sure it is inserted last in the mastering chain. You are going to use the Limiter plug-in to maximize the loudness of your track, while at the same time preserving transients and minimizing limiting artifacts like distortion and breathing-and-pumping. The limiter has three modes that control the speed of its auto-release scheme: fast, medium and slow. If you are aiming for a competitive master, you want the auto-release to be as fast as possible without generating distortion artifacts. However, achieving this can sometimes be difficult when large amounts of gain reduction are necessary. In that case you might have to compromise between a grittier but loud master, or a smoother but less loud master - either by choosing a slower auto-release, or by manually taking back the threshold a little. On the other hand, if you are aiming for a more conservative master, and therefore less gain reduction is needed, a medium or slow auto-release is usually more appropriate. This will require a slightly lower threshold to compensate for the slower auto-release, but will overall produce a smooth-sounding master without any distortion. Remember though, that in addition to decreasing loudness, an auto-release that is too slow might cause breathing-and-pumping, especially in case of high amounts of gain reduction. Go through the following steps:

► To get a feel for the Limiter effect and its artifacts, set the mode to fast, the ceiling to 0.0dB and turn off the maximize control. Gradually lower the threshold from 0.0dB to -48.0dB, while watching the gain reduction meter. As the maximize control is turned off, the plug-in will essentially stay inactive, until the threshold gets low enough to catch the first peaks of your track. The meter might show a few dB of gain reduction, but the audible difference will be negligible. When lowering the threshold further, causing the plug-in to generate more and more gain reduction, you will start to hear increasing distortion and breathing-and-pumping artifacts. Skip through the fast, medium and slow mode, and switch between the different transient settings, to get a good feel for the Limiter's inner workings, and to clearly identify the artifacts the plug-in produces with different settings when pushed hard.

► To have a real-world loudness reference, put your reference tracks back to their full volume. Take a moment to compare your track to them, to get a good idea of the target loudness you are after. Then, put the Limiter's threshold back to 0.0dB, turn on the maximize control and set the ceiling to -0.3dB to compensate for intersample clipping and the potential volume boost created by converting to MP3 later on. Then, play the loudest part of your track, and slowly lower the threshold. As the maximize control is turned on, the volume of your track will now increase automatically. Try to reach your target loudness, without causing audible distortion and breathing-and-pumping. Experiment with the different auto-release modes and transient options, to find a good compromise between loudness and quality, while preserving or even boosting your track's transients.

Looking at the gain reduction meter, you will probably find that you can get away with about 3.0dB to about 6.0dB of gain reduction without doing too much harm to your track. If you are aiming for a conservative master, this is usually enough. Generating more gain reduction however, will quickly cause audible artifacts. If you are aiming for a competitive master, but you can't reach your target loudness without doing harm to your track, you can make a few strategical tweaks to your mastering chain:

Step 1 use the Monomizer to tighten up the low end stereo image some more, creating a more stable foundation for the other steps.

Step 2 use the Equalizer to cut some lows with a shelve, creating more headroom for the Compressor and Limiter.

Step 3 Use the Compressor to even out the dynamics of your track some more with the compression for impact technique, setting the saturation to hard. You might have to do some subtle damage at this point to give the Limiter in step 5 the necessary headroom.

Step 4 Use the Convolver plug-in to add more of the wet convolved signal, increasing the density of your track, and the additional Equalizer to boost the frequencies important for psycho-acoustic loudness.

Step 5 Lower the threshold to see if you can achieve your target loudness after tweaking the other steps in the chain. If you are still short a few dB, insert a additional instance of the Limiter before the current instance, set its mode to fast, and use it to pre-clip the highest peaks in your track.

When aiming for a competitive master, usually you can't avoid a compromise between loudness and quality. However, using a combination of techniques that increase both psycho-acoustic as well as actual loudness (and perhaps trying a few different brickwall limiter plug-ins), will generally get the job done. Once you have chosen your Limiter settings, you are ready to render your master as a stereo .WAV or .AIFF file of 44,1kHz and 16bit. However, if you are working on 24bit, it is advisable to first add a dither signal to your track, to avoid any low level quantization distortion, caused by lowering the bitdepth. Such a dither function is standard in most sequencers and audio editors. After rendering, trim the silences and create fades at the start and end of your track, and you are done mastering your track.

In conclusion

Having gone through the whole mastering process, it is a good idea to review your approach and the results you have achieved in order to learn as much as you can from the experience. This is best done by playing your mastered track on many different audio installations to see if it holds up. Remember though, to always start by playing your reference tracks first, to calibrate your brain to a new room and playback system. It is deceptively easy to fool yourself if you don't. Hopefully you are happy with your track, and you now realize that mastering is not some kind of dark art, but instead a set of specific skills that can be studied and practiced. In that regard, the biggest challenge in mastering your own music in your home studio is understanding the mastering process and developing a mastering workflow that suits you. You need this understanding and methodology to be able to trust your ears and make clever decisions during any mastering session, but don't forget that good taste and artistic vision are just as important. Once you have found a suitable workflow and know how you want your music to sound, mastering can quickly become a second nature. So, learn and experiment all you can, become opinionated and develop a workflow of your own. Good luck mastering!

For more information about the book Home Studio Mastering,
go to www.homestudiomastering.com.

Want to learn more?

Check out the full book at
www.homestudiomastering.com
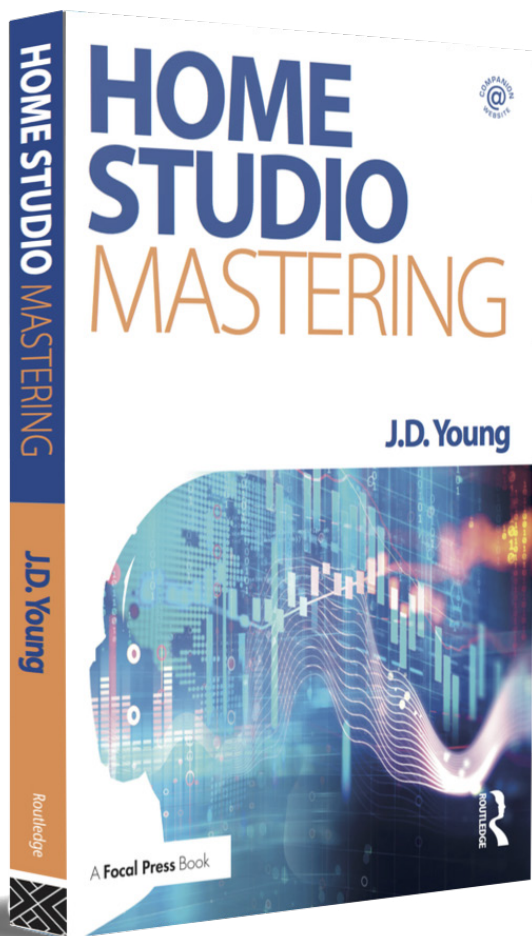
Home Studio Mastering is a step-by-step manual that gives you all the tools to professionally master your music yourself. It demystifies the subject in a hands-on way for those working in a home studio and provides comprehensive guidance, from buying equipment and applying acoustical treatment, to using different audio applications and mastering plug-ins.

► Set up your own studio
► Explore mastering techniques
► Learn a mastering workflow
► Look up mastering terms

A step-by-step guide
for mastering your music
in your home studio