tonebase Academy | Recording Course

III. All about Dynamics

ACADEMY

Your Instructor: Martin Zimny

Suitable For: All Levels

A tonebase Workbook

"Take your recording to the next level, technically as well as aesthetically!"

Prior Knowledge

- The basics of Repear as covered as in the first two lessons

Contents

Lesson Summary

Let us dive deep into our recording and elevate it with different signal manipulators. We will look at level-based tools like a compressor, frequency-based tools like an equalizer, and time-based effects like reverb. Additionally, we will ensure maximum playback level by normalizing our output volume with a limiter.

What You'll Learn:

- How and when to use compressors and equalizers
- How to add reverb to your recording
- How to export a professional-sounding audio file

Before we Start

[CLICK TO WATCH THE LIVE SESSION (1/25/21 @ 11 AM PST)]

[CLICK TO VISIT THE FORUM THREAD FOR WEEK 3]

There are no hard-and-fast laws in audio post-production. I have used Melodyne Studio to correct a wrong note within a chord in a multitrack recording because booking another day in a hall for $500 to fix one note isn't worth it. I have used parallel distortion to saturate the sound of a dull guitar. I have used a De-Clicker to get rid of the valve sounds of a flute. At one point, I was wondering whether I could code a plugin called the "David Russ-ler" that adds weight to your sound by parallel compressions, boosting warm frequencies, and cutting harsh sounds with a high-shelf filter.

This being said, you should typically make as few manipulations as possible to maintain the integrity of your source. Not only is a classical musician quite unhappy if they can't recognize the sound they've trained their whole life for, but you are also actively furthering the sonic degradation of the signal by adding plugin after plugin. In this lesson, we will focus only on those alterations that enhance the listening experience. These include taming hard peaks in your recording to maximize the listening quality, low-cutting your signals to prevent ultra-low bass rumbling, and maximizing your signal to ensure correct playback volume. We will work with three different signal manipulators: the compressor, the equalizer, and the limiter. We will also introduce a subtle reverb, since classical music is usually heard in large concert halls.

Historische Stadthalle in Wuppertal, Germany

Manipulating Dynamics with Compressor and Equalizer

For this course, we will only work with Reaper's Stock Plugins. If you have a very nice equalizer or a lush sounding reverberator, feel free to use that. Keep in mind that before we expand our toolbox, it's often helpful to understand what we already have.

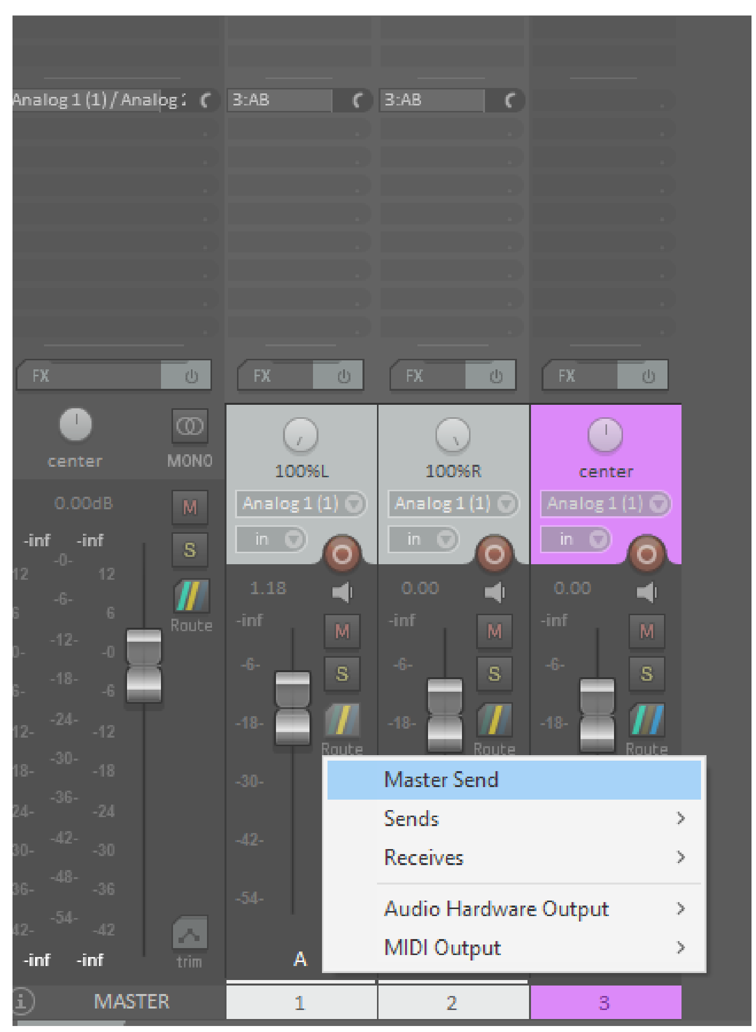
Equalizers and compressor do almost the same thing, but with important differences. While the equalizer reduces the level of your signal at a particular frequency range, the compressor reduces the whole spectrum according to the input level.

Routing your Signals Together

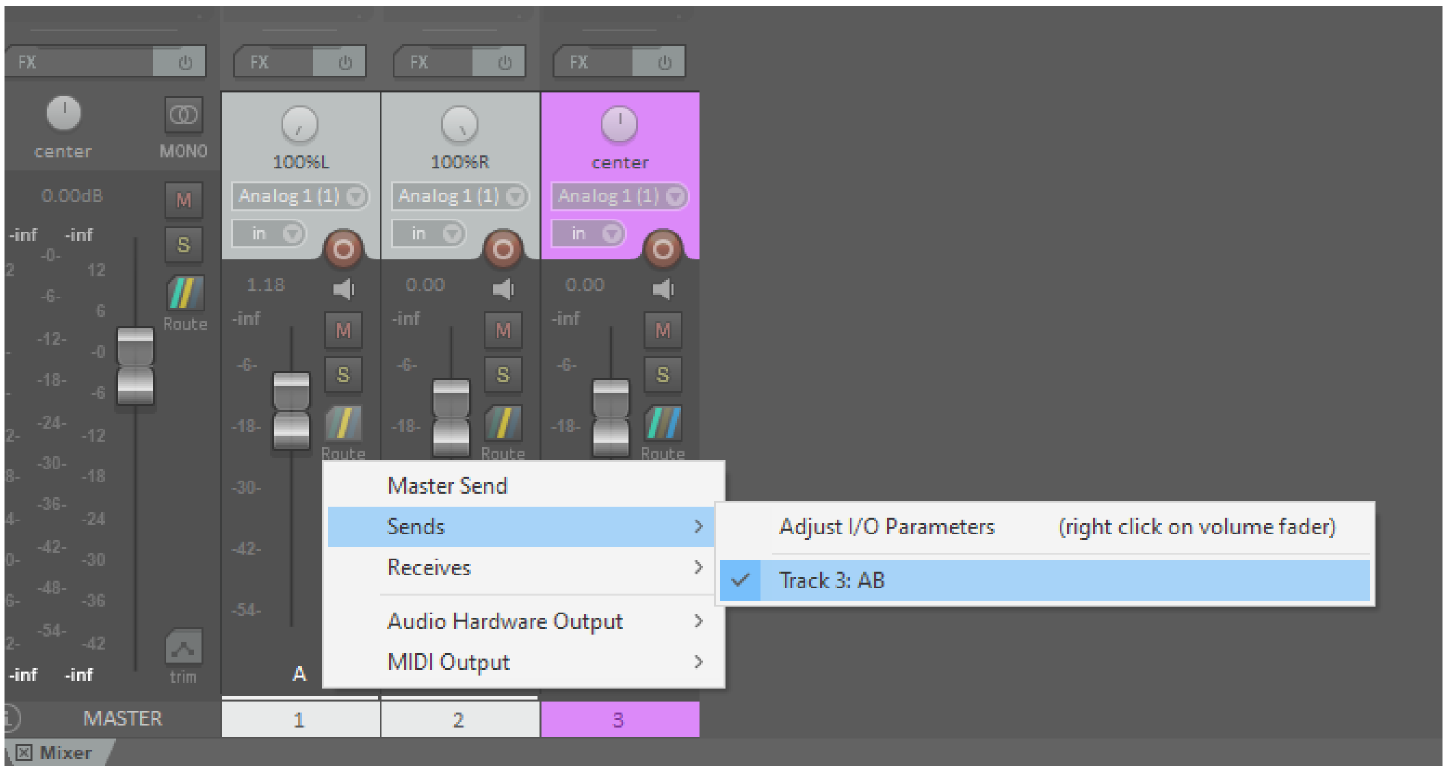
Depending on your setup, you want to first route your signals together. I recorded two separate mono signals, so I want to create a stereo track that combines both signals. This way, I can manipulate them as a unit. If you already recorded onto a stereo track or only recorded one mono signal, you can skip this step for now (although we'll use routing when we apply reverb).

First, create a new track by hitting the shortcut ctrl/command-T or by double-clicking in the track header area. Give that track a name (AB or Stereo Sum) and optionally, a color.

Then, head over to your mixer area, select your two mono channels, and left-click on "Route". Deselect "Master Send".



Send your mono channels to your newly created track:



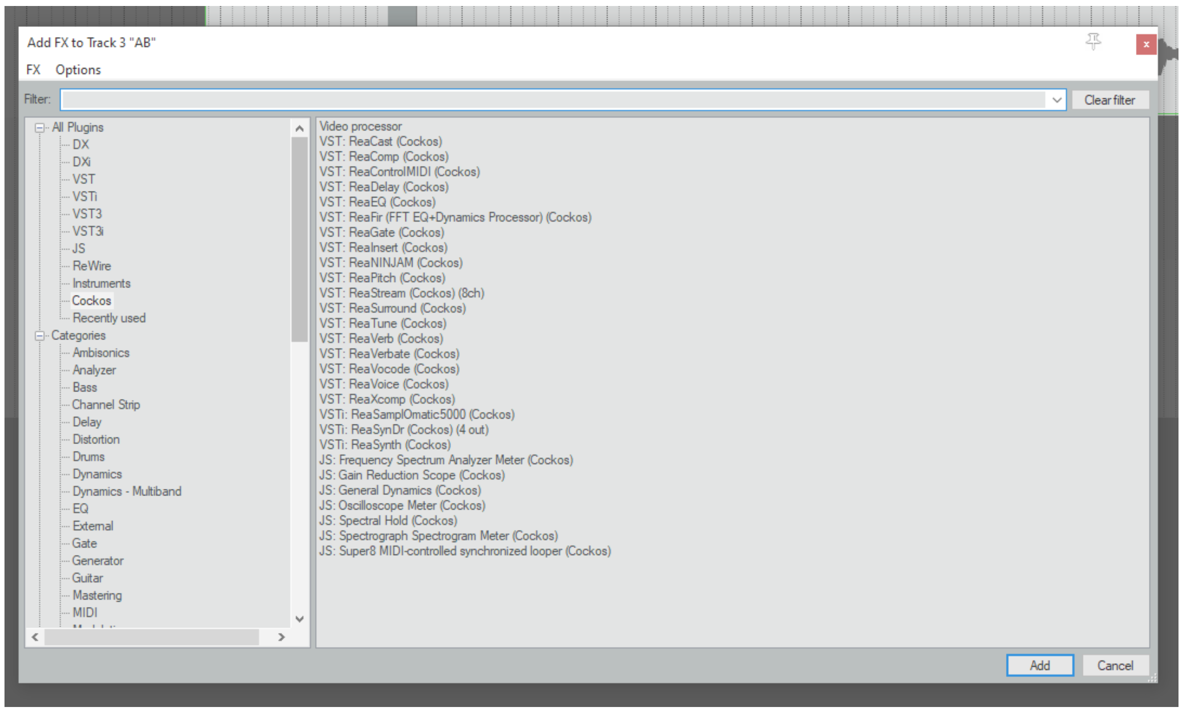
Make sure that the new track has "Master Send" activated. The different colored stripes on the "Route" button represent what function the track has: Is it sending? Is it receiving? Is it sending to the master track? Now that we created a new track that sums together our input mono signals, we can apply our compressor!

The Compressor

Compressors are used infrequently in classical music since we want to retain as much dynamic contrast as possible. Over-compressing a signal results in a non-engaging listening experience or active "pumping" of a signal if the compressor is misused.

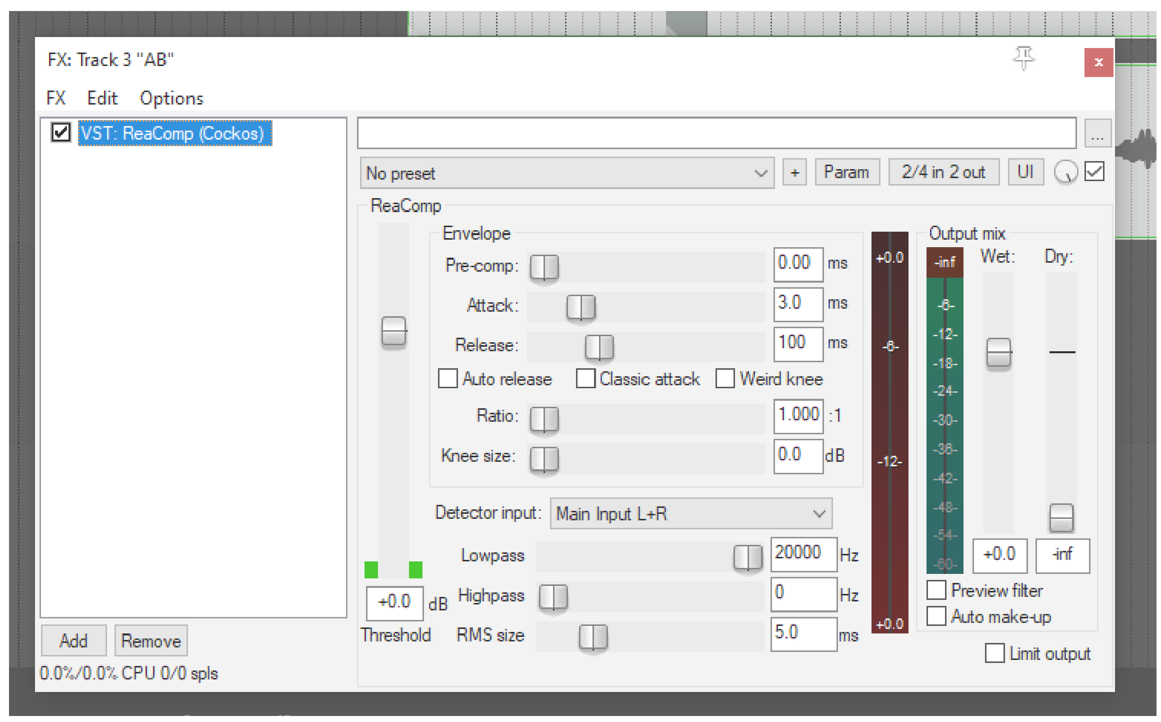
Insert the compressor by clicking on the upper half of the empty fields of your corresponding mixer channel.

The following FX Window should open up:



This is the hub for all the effects that live on your machine. As we search for our compressor, you can use the filter field and type in "ReaComp”. If you double click on it, the filter will be added to your channel and the compressor window will open.

For our purposes, we only need to worry about threshold, attack, release, and ratio:



Threshold: Defines the volume level where the compressor will start to act.

Attack: Defines the delay time in milliseconds when the compressor starts to work. Sometimes you want to delay the compressor and allow the transients (harsh attacks) to pass through.

Release: Defines the time the compressor takes to be back at zero gain reduction.

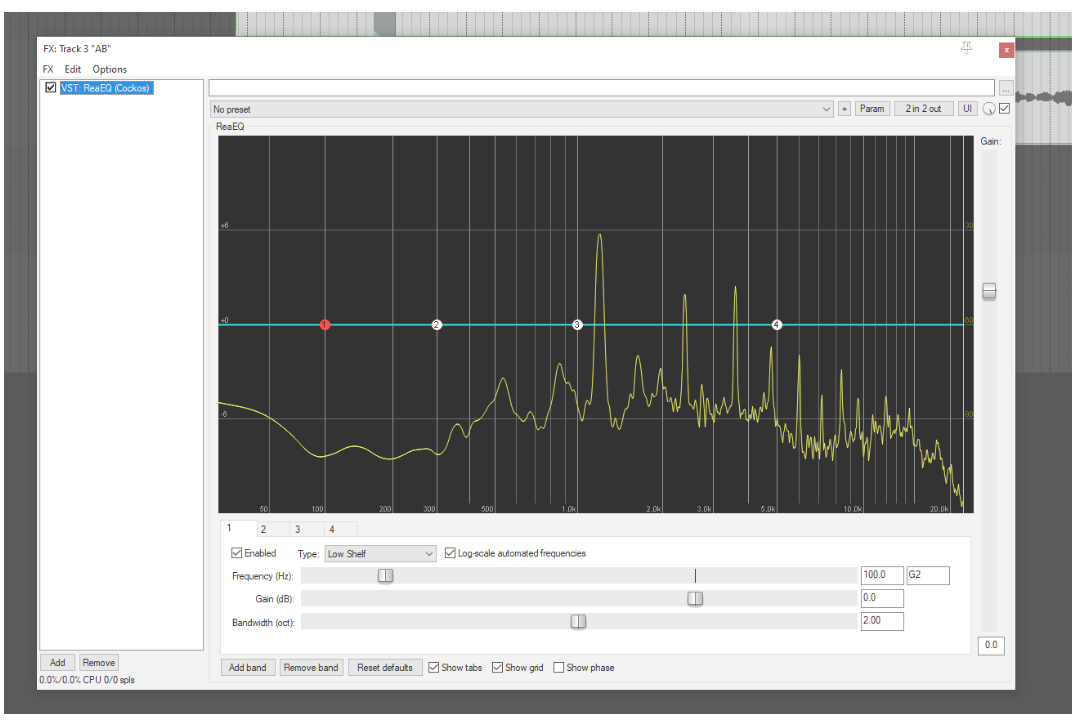
Ratio: Defines the amount of gain reduction of the signal once it passes the threshold.

Next, I would navigate to the loudest passage of my recording and set the ratio at around 1:1.5 or 1:2, slowly reducing the threshold until the compressor starts working (in pop music you would go much higher, but we want to retain as much dynamic as possible).

A ratio of 1:2 with a threshold of -25db means that every decibel above the threshold will be reduced by half. So if a signal is -24db, it is 1db above the threshold, and this decibel gets reduced by half, resulting in a signal of -24.5db. This may not sound like much, but this is exactly what we are going for.

If I have set everything correctly, I would go for about 3-6db gain reduction *maximum*. This is highly dependant on your signal; if everything is always very loud and has a lot of high peaks, I would probably *not compress at all*. If I have a very quiet piece and there is one peak that costs me a lot of overall loudness, I would go for an even higher gain reduction.

The Equalizer



After you compressed your signal, you can add your equalizer. There is a lot of discussion as to whether to add an EQ before or after a compressor, but as we typically only do minor adjustments in classical music, this doesn't matter too much. In short: *compress first, all the others do it wrong!* (Jokes aside, the compressor is input-dependent, meaning any EQ we add *before* the compressor will change the way the compressor behaves, making it harder to predict).

Insert the EQ the same way you inserted the compressor by opening the FX window, but now look for the plugin "ReaEQ".

Using an equalizer is pretty straightforward. First, choose a frequency band and make adjustments to your signal. Typically, it's a better idea to *cut* frequencies rather than boost them, as raising the volume could accentuate phase problems that may exist in your recording.

There are typical EQ adjustments that I make for classical guitar: since the 6th string in regular tuning is the low E at about 82 Hz, we won't do too much harm to the signal if we cut the signal at around 40-50Hz with a **High Pass Filter**. This reduces any low rumble that transmits through the floor into your microphones like street noise, ACs, etc.

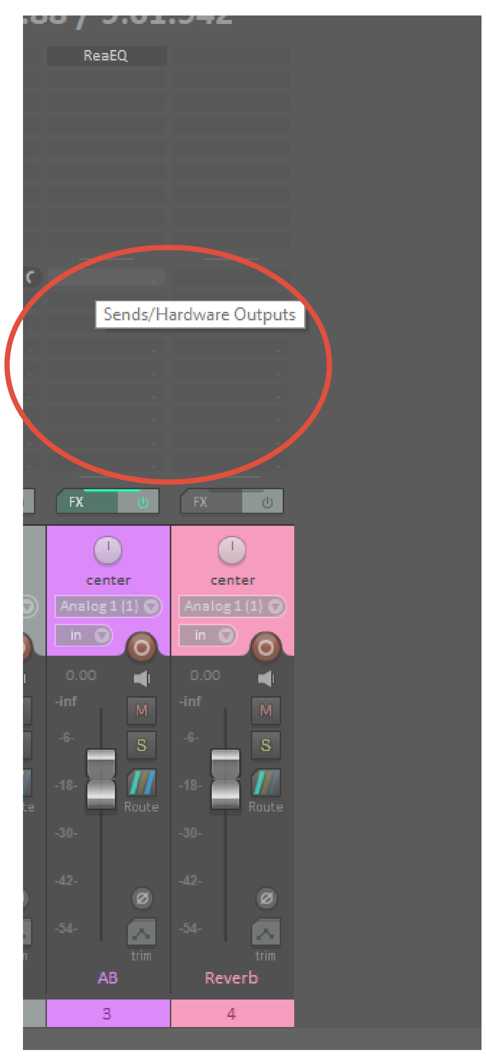
Afterward, I would increase the fidelity of my signal by boosting 2-3 dB around 8.5khz with a **High Shelf Filter**, just a little bit higher than any nail noises. If you experience a certain "boxiness" in the sound of your guitar, you could work between the 400-800Hz area and cut your signal in small increments.

A lot of small steps go a long way! Play around a little bit, but don't go crazy with your signal. It's never better to fix something in post-production that could have been improved by a better microphone placement.

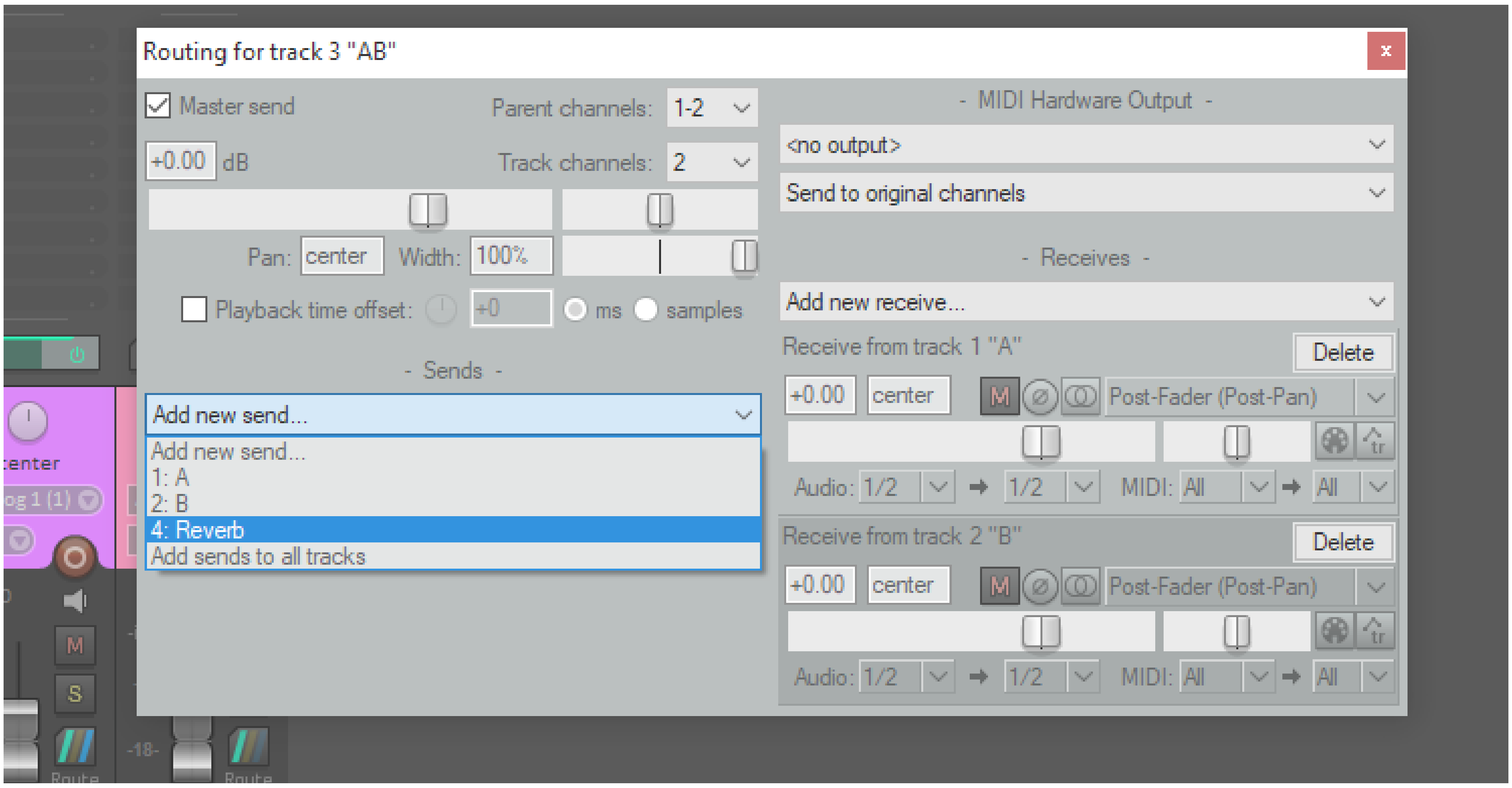
Reverb

Time-based effects like reverb and delay work a little bit differently than EQ and compression. Since they need more CPU, it's usually a good idea not to put reverb on the audio track itself. Instead, we create an extra track and *send* our audio signals to it.

To create a send, click anywhere in the "Sends/Hardware Outputs" section of the channel strip.

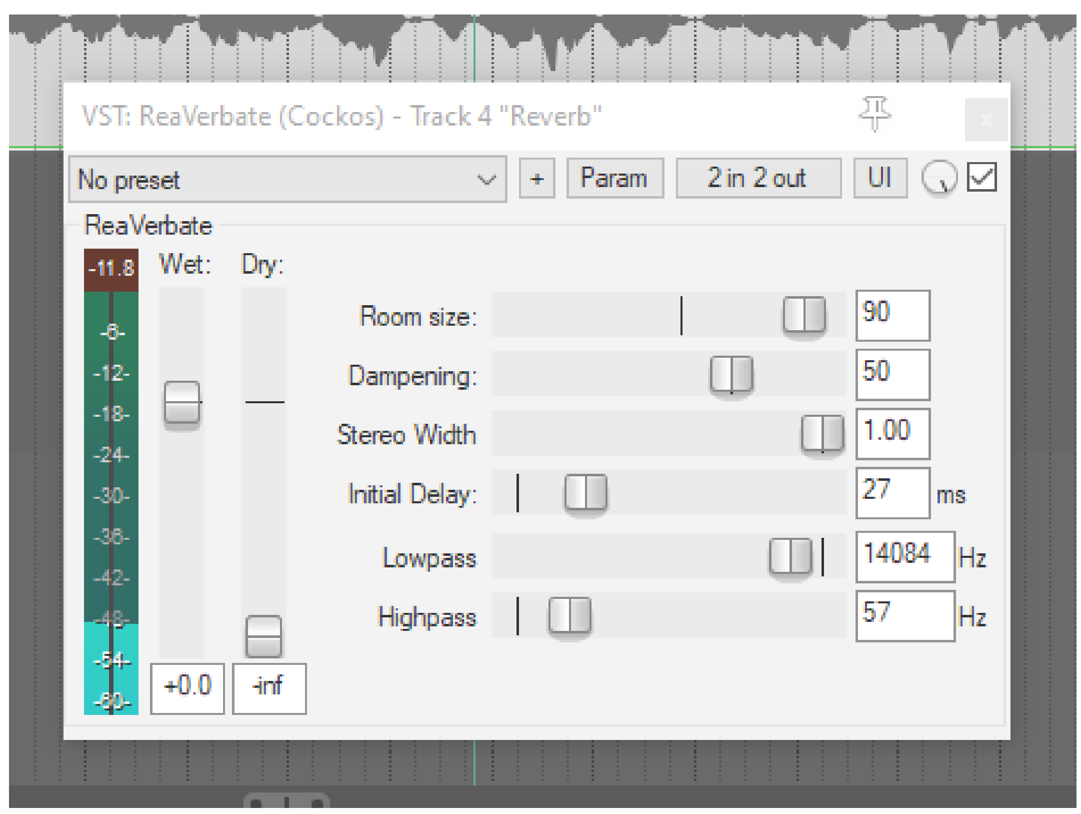


When the routing window opens, add a new send to the newly created track by clicking **"Add new send"**. I named mine "Reverb".

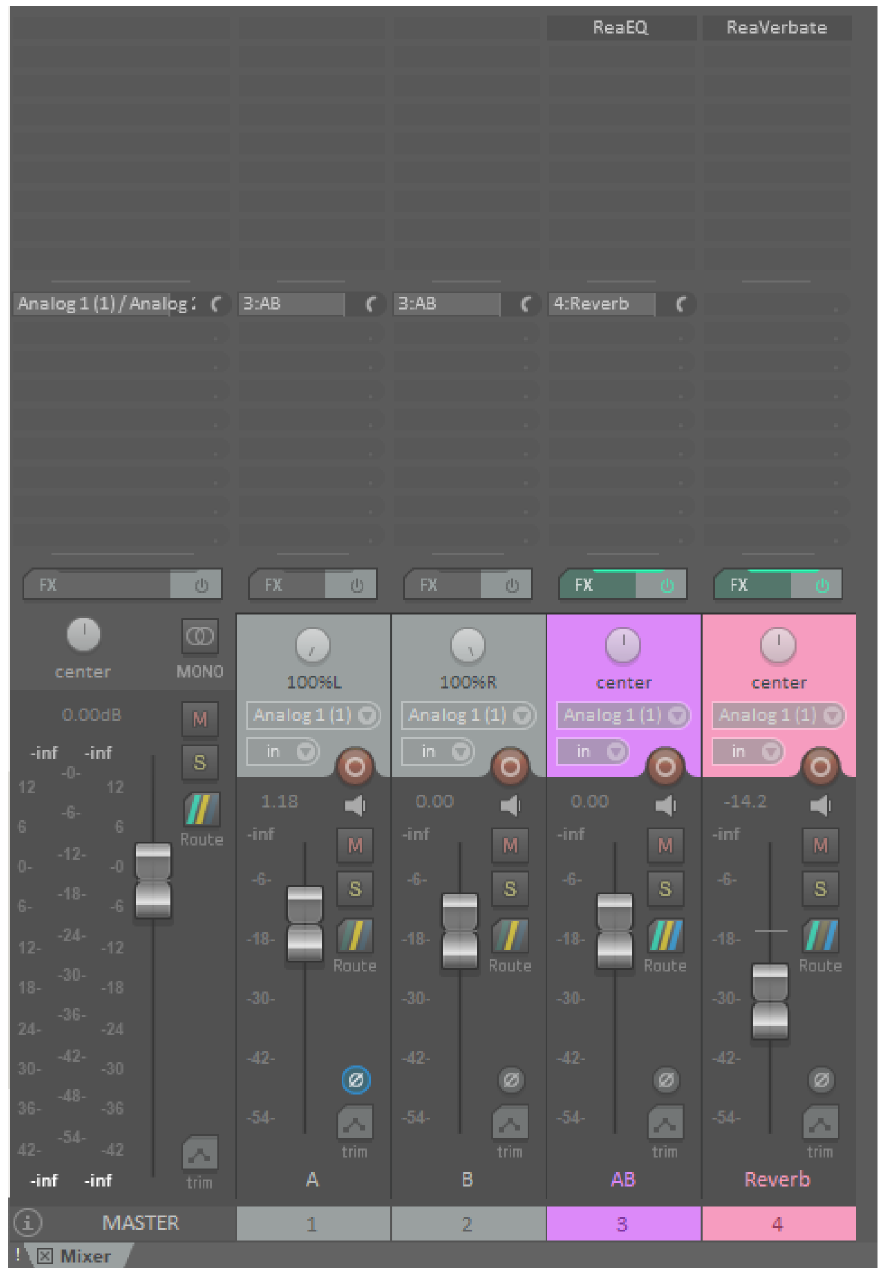


Next, add the reverb effect to your new track the same way you added the equalizer and the compressor. Search for "ReaVerbate" in the effects window, the stock plugin for reverb.

The first thing to change is the wet/dry ratio. We don't want any dry (unaffected) signal on this channel, since that's already coming from the audio track. The wet signal is the output that contains reverb. Set Wet to "0db" and Dry to "-inf". Then, play around with different parameters to make the reverb better fit your needs.



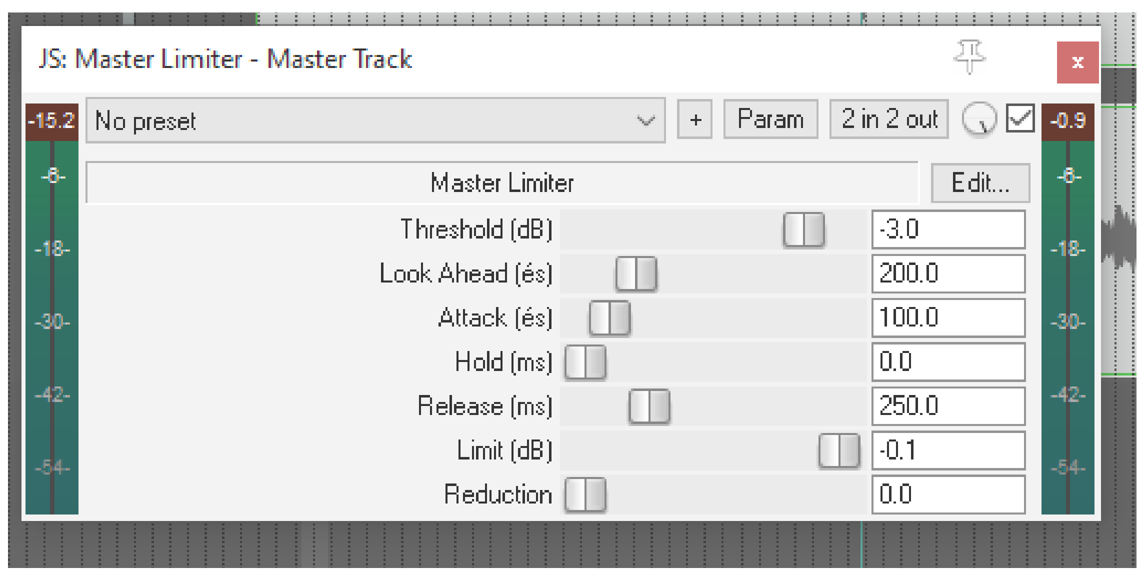
This is quite a strong reverb, but since it lives on its own track, I can always reduce its level through the corresponding fader in the mixer.



Limiting

We are almost done with the mixing stage of our recording. The last thing to do is to ensure our tracks are playing at the correct level. For this, we use a limiter, which is essentially a compressor with a 1:inf ratio.

We will add an effect to our master track, the "MasterLimiter". The process is the same as the other effects, only on the master track.



The most important parameter here is the threshold fader. It's important to keep the right metering in sight as this is showing your final output. For this to work properly, navigate to the loudest part of your recording and hit play.

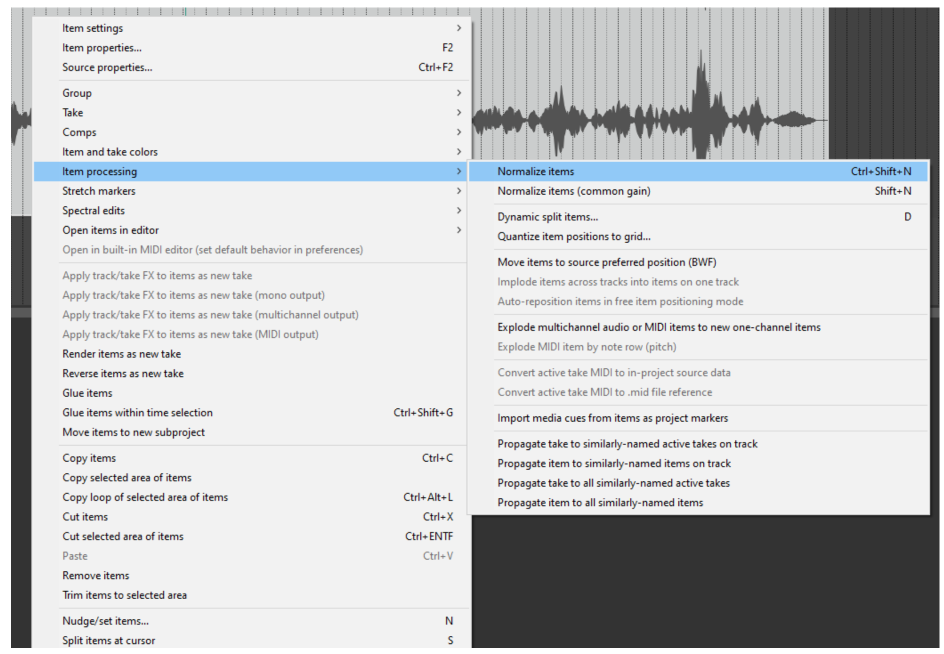
Then, slowly lower the Threshold fader (the position at which the limiter starts to affect the signal). Your limiter is set properly if the output shows -0.1db and the reduction fader is still.

Personally, I find Reaper's built-in limiter to be somewhat clunky to work with. On the next page, I'll show you an alternative process involving the Normalize feature.

Export and Normalize

To begin, select your entire recording. Then, go to **File→Render** and export it as a WAV file to ensure maximum quality for further post-production. After you export it, open a new project tab by hitting the shortcut ctrl/command+alt+N and drag-and-drop your newly created WAV file into the empty project. This will create a new track containing your WAV file.

Right-click on your item and select **Item Processing → Normalize Times**. This will set the loudest peak of your recording to 0db. To ensure no problems with MP3 playback and encoding, it is better to be between -0.1 and -0.3db. You can adjust your output volume by lowering the fader in the mixer to -0.1db.



Re-export the file again, and congratulations! You've just created a professional sounding WAV file that would be suitable for uploading on streaming services like Spotify, Apple Music, and more! All streaming services have different recommended file types, but we're off to a great start!

Assignments

- ○ Take your recording to the next level by subtle use of EQ, Compressor (if necessary), Reverb, and Normalizing.
- ○ Take screenshots of the plugins you used and upload your audio to the forums.

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CLICK TO VISIT THE FORUM THREAD FOR WEEK 3

CLICK TO REVIEW THE RULES AND FAQ

Questions

Use the questions below to check your understanding of the material from this session. Use the dedicated forum thread to post further questions or discuss topics from this workshop. Martin will post the answers on the forum a couple of days after the live stream takes place.

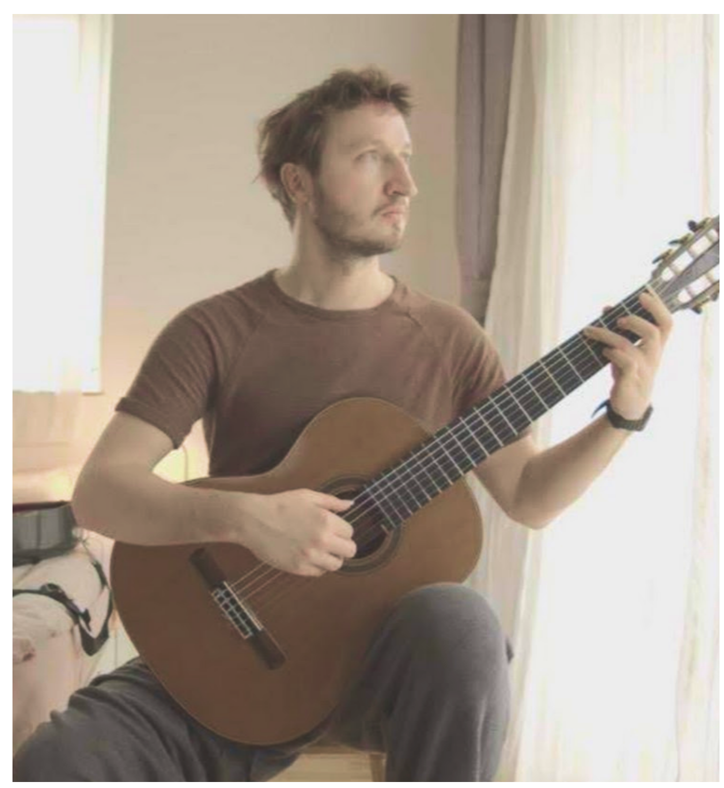
1. What is normalizing?

2. How do we reduce low bass rumble?

3. What is displayed in the EQ?

About Your Instructor: Martin Zimny

Martin Zimny, born in 1988 in Munich, Germany, graduated with a Master's of Music from the Robert Schumann Hochschule in Düsseldorf, Germany with Cuban guitarist Joaquín Clerch. He has won prizes in several national and international competitions and played concerts across Europe and India. He has taken part in festivals and workshops in Austria, Germany, Spain, the Netherlands, and Serbia. Martin has been working as a guitar instructor for almost 10 years. After his degree in music, he studied Engineering for Audio and Video at the University of Applied Sciences in Düsseldorf. Today, he continues to perform and teach while working as a professional recording engineer.



Notes