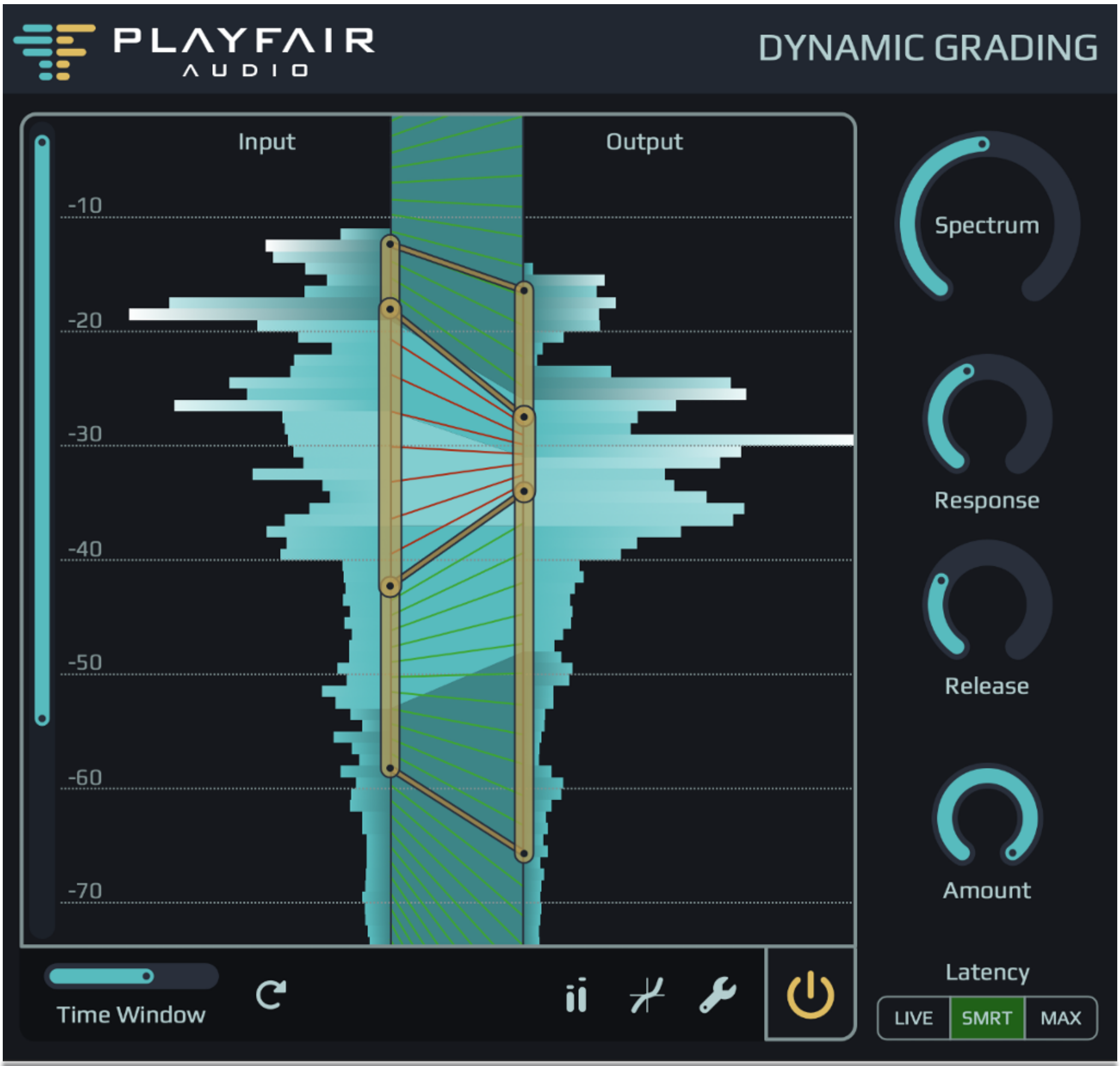


Dynamic Grading

User Manual

Contents

About Dynamic Grading

Dynamic Grading is an entirely novel approach to dynamic processing of audio. Unlike traditional dynamic processors like compressors or expanders, which are notoriously hard to adjust, Dynamic Grading lets you mix with more confidence than ever before. It does so by showing you an easily comprehensible histogram of audio dynamics, providing you with intuitive and easy-to-use means for manipulating it graphically, and using sophisticated and musical dynamic processing algorithms.

The Dynamic Grading method is inspired by digital image processing tools, where the use of histograms and grading curves is well-known and established to adjust the dynamics of brightness in images. With Dynamic Grading, these concepts have been adapted and tailored to practical use in music production and audio engineering for the first time.

Our plugin is designed to make mixing a song, adjusting movie dialogue or podcasts and mastering albums an intuitive, fast, and enjoyable experience, as easy as drawing a stick man.

Installation and Activation

Playfair Audio uses iLok by PACE Anti-Piracy Ltd. for copy protection and license activation. To use our software, you need to have an account on iLok.com and have the iLok License Manager installed. If you have not used iLok before, please go to iLok.com to sign up and download the License Manager.

To install Dynamic Grading on your operating system, open the disk image or ZIP file you downloaded from our website, double click the installer file ("Dynamic Grading.pkg" on macOS, "Install Dynamic Grading.exe" on Windows) and follow the directions on screen.

After installation, Dynamic Grading will appear in the plugin list of your favorite Digital Audio Workstation.

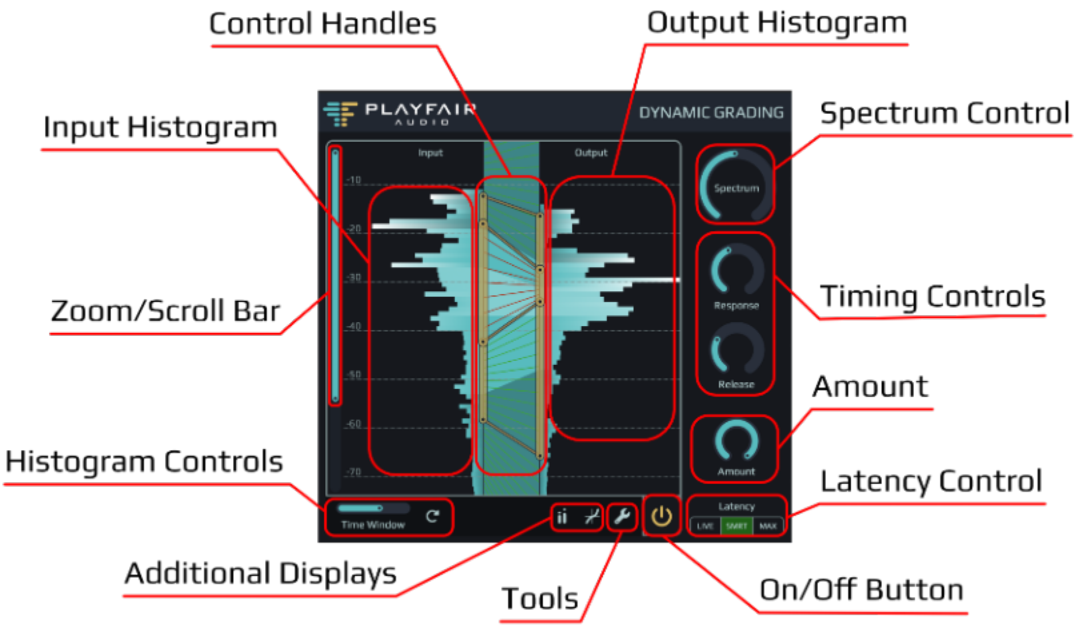
When loading the Dynamic Grading plugin for the first time, you will be prompted to activate the plugin. Follow the instructions on screen to either start a trial license or activate your purchased perpetual license.

The trial license is fully functional and will expire after 14 days. To continue using Dynamic Grading after trial expiration, you will be prompted again to enter your activation code, which is available for purchase on our website or at authorized resellers. You can also use the iLok License Manager to enter your activation code and manage your licenses.

The Dynamic Grading license can be stored on an iLok USB key, the iLok Cloud or your local computer. For more information on using iLok and the License Manager please refer to iLok.com.

Basic Operation

Overview



Dynamic Grading's user interface is built around the main display, which shows the dynamic histograms for the input and output signals. The area between the two graphs provides graphical control handles to intuitively adjust the dynamic processing.

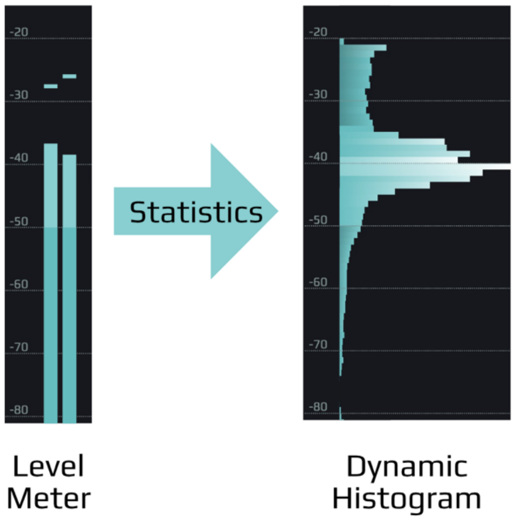
The Zoom/Scroll bar to the left lets you adjust the visible dynamic range, either by moving the bar like a normal scroll bar, or by moving the endpoints to adjust the visible range. Alternatively, you can scroll using the mouse wheel, and zoom in/out by pressing the Alt key while using the mouse wheel.

Below the main view are controls for adjusting the histogram visualization, toggling additional displays, and accessing other options. An on/off button lets you bypass processing.

The controls to the right affect the measurement and processing algorithms and can be adapted to the specific signals you are dealing with in a meaningful and musical way.

Understanding Dynamic Histograms

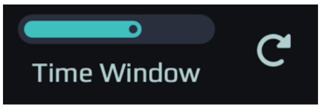
In Dynamic Grading, audio dynamics are displayed as so-called dynamic histograms. These graphs represent a **statistic of perceived momentary loudness over a given length of time**. The easiest way to think about it is via the commonly used level meters you know from DAWs and mixing desks:



Now imagine a *really fast-moving* level meter like the one on the left, and a superhuman statistician sitting in front of it, looking up the level on the meter *really often* and keeping a tally sheet of how often she or he reads each possible value over – say – the last 20 seconds. The result is plotted as a bar graph like the one on the right. In the example above, you can see that values around -40 dB have occurred much more often than values around -50 dB or -16 dB, as an example.

This kind of graph tells us a lot about the dynamics of audio recordings. It reveals the dynamic range (the highest and lowest readings) as well which loudness regions are most prominent and how much perceived loudness varies (= how dynamic it is).

> *A sharp peak in the graph hints at a static tone, note or noise with a constant level, while a broad and flat shape means there is a lot of dynamic variation.*



You can adjust the length of time considered in the analysis using the **Time Window slider** on the bottom left of the main view. For best results, the time is set in bars relative to the tempo and time signature of your song.

> *A longer time window leads to a more stable and smoother histogram. A shorter time window will react faster to adjustments or a significant change in source material, such as upon a transition from verse to chorus.*

To restart the histogram computation for faster adaption after changes to the source audio, click the "Clear" button.

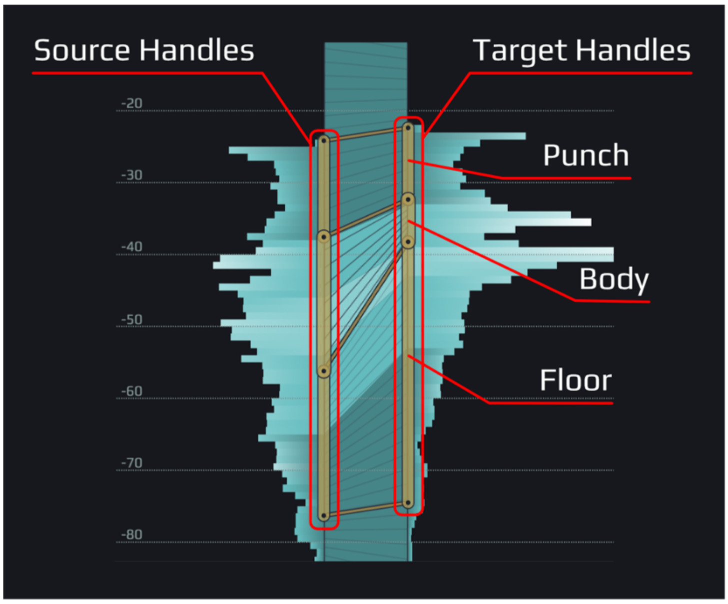
Additionally to the long-term histogram, the brightness of the bars in the graph shows the short-term loudness.

> *By **listening and observing** when which bars light up, you can determine which sounds in your source material contribute most to which parts of the dynamic histogram.*

This is easier to understand than it is to explain. By using Dynamic Grading on different audio material and observing the resulting graphs, you will quickly understand how to read it.

The Dynamic Grading Process

Now that we've understood how to read dynamic histograms, we can start manipulating them to our liking. Dynamic Grading makes this easy by providing graphic controls that let you determine important regions of the source material's dynamic range and shape these regions to where you want them to be.



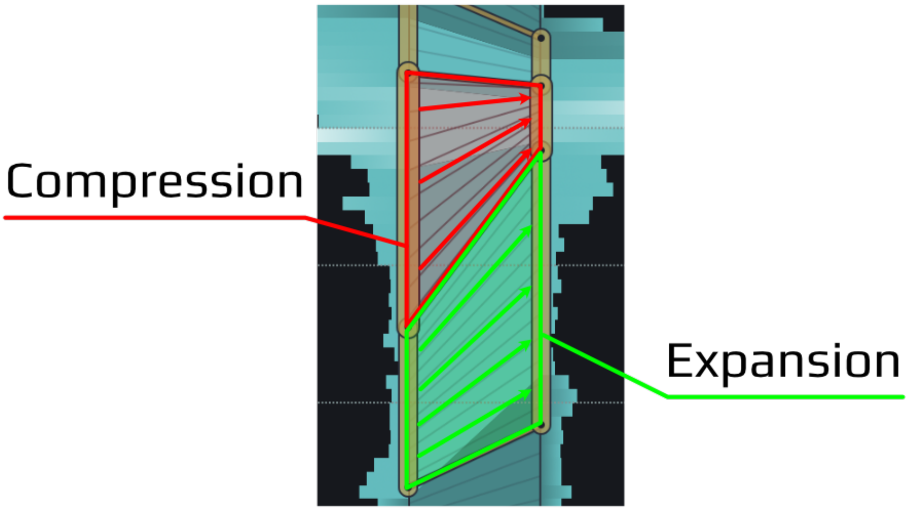
The yellow shapes in the middle between input and output histograms can be freely manipulated by dragging them around with the mouse. The handles on the left (input) side are called source handles, the ones on the right (output) target handles. We call the yellow bars "ranges", and the little dots "reference points".

Both sets of markers consist of three distinct regions that you can use to shape audio dynamics. **By adjusting the source handles, you specify where the punch, body and floor ranges are in the source material.**

Listening to the audio and observing the resulting dynamic histogram will quickly give you a sense of what that means. For most practical audio signals, you will recognize a **body** range where the "meat" of the signal is located, such as **vocals or note sustains** in instrument tracks. Above that, in the **punch** range, you will mostly recognize **transients, consonants, percussive elements and note onsets**. The **floor** range on the other hand usually contains things like **late decay of instrument notes, reverberation, or background noises**.

> ***TIP:** When starting from scratch, first move around the connecting lines instead of the source handles alone. That way, you can set up the source regions according to the histogram reading, without affecting the audio yet.*

Once you've set up the **source handles**, adjust the **target handles** to specify where you want the body and tail regions to be. Depending on how the source and target ranges are set, you get a different type of dynamic processing in that region.



> *When the target range is narrower than the corresponding source range, this part of the dynamic range will be compressed. If it is wider, it corresponds to an expansion of dynamic range.*

For example, you will most of the time want to compress the **body** range to make it more narrow, so the most prominent features of the audio track sit nicely in the mix without competing too much with other tracks.

By adjusting the **punch**, you can then tightly control how prominent onsets, transients and percussive elements should be relative to the body range. Narrow the punch to get smoother onsets, widen it to get more punch.

With the **floor**, you can adjust reverberation and background noise. Widen the floor to smoothly reduce reverberation and noise, or squeeze it for weird and exaggerated artistic effects.

Below is a handy cheat sheet to summarize the typical anatomy of audio signals and what you can do with them.

Range	**Contents**	**Compress to…**	**Expand to…**
Punch	Note onsets, transients, attack	Tame peaks, move to background	Increase punch, move to foreground
Body	“The Meat”: notes, sustain	Increase presence in the mix	Create space, enhance „groove“
Floor	Reverberation, noise, echoes	Exaggerate reverb, artistic effects	Dereverberate, reduce noise

Advanced Handle Editing

The way you edit the graphical handles is designed to be mostly self-explanatory and tailored to practical use. Just try it out, and you'll quickly get used to it. You can **drag the individual reference points** around, or **move the punch or floor ranges** as a whole. You can also **move the connecting lines between reference points**, which is especially recommended when you're starting from an initialized setting and want to initially set up source ranges while keeping the neutral setting.

The source body range moves all handles together. This is useful when the **input level** has changed (e.g. because of modifiactions to another plugin before Dynamic Grading). Similarly, the target body range by default moves all target markers, which changes the **output level** in a neutral way. **To move only the respective body ranges, hold the shift key while dragging.**

When you **hold the alt key while dragging a range**, you can squeeze or stretch said range while keeping the center position.

When you **hold the alt key while moving a reference point**, it will be locked at the position where the respective range is linear, which means it's neither compressing nor expanding.

Similarly, **double clicking either ranges or reference points resets them to a neutral position**. Double click the target body range to achieve a neutral setting while keeping the source handles. Double click the source body range to initialize all handles to the default setting.

The Spectrum Parameter

Dynamic Grading's algorithm is designed to accurately reflect the human perception of audio dynamics. In this context, the frequency spectrum, and especially the **balance of higher and lower frequencies in the source material** is important to get musical results.

The Spectrum parameter lets you adapt the measurement to the spectral balance of your source material. Most real-life music signals approximate a so-called **pink spectrum**, where the energy towards higher frequencies decays with 3 dB per octave, which is the default and center of the spectrum parameter.

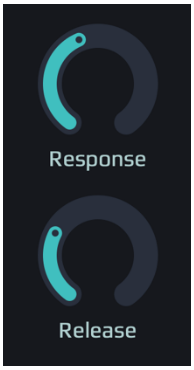
> *Turn the Spectrum parameter to the left if your source material is closer to a white spectrum, which is perceived as more treble-heavy, or turn it right if your source material leans strongly towards more bass, which is called a red spectrum.*

In practice, **it is easiest to adjust the Spectrum parameter after adjusting the source and target ranges** so you hear some audible processing already. Then adjust the Spectrum parameter to a value that sounds most natural and musical to you.

A good way to explore the effect of the Spectrum parameter is by squeezing the body range and high tail on a drum bus and paying attention to how the kick and snare react to the processing. In extreme white or red settings, the kick and snare will often react very differently, while a setting closer to the pink spectrum will usually lead to a much more balanced and natural sounding effect.

Note at this point that most traditional compressors work in a way that corresponds to the full white spectrum setting, which is something all audio engineers should be sad about.

Timing Parameters

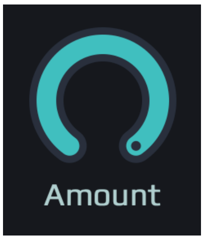


Dynamic Grading uses two timing parameters to determine how fast the dynamic processing acts: Response and Release.

The Response time determines the time over which loudness is evaluated for building the dynamic histograms and for dynamic processing. Choose longer Response times for smooth processing, and short Response times for a more drastic and fast-acting effect.

The Release time affects how the dynamic processing recovers after a significant sound event. It is especially useful in conjunction with fast Response times to keep note decays and reverb tails natural even under drastic grading settings.

The Amount Parameter



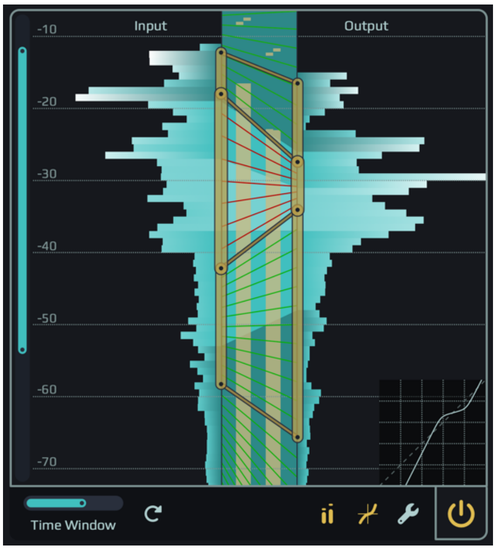
Sometimes you may find yourself gone a bit too far. And you may find yourself thinking that setting is pretty good, but just a little bit too extreme. And you may ask yourself: “Well, how did I get here?”

We don’t know how you got there, but the amount parameter lets you easily dial back the amount of processing. That way you can access all the settings between where you are, and where you came from (no processing).

Note though that this “dialing back” is not reflected in the handle positions of the main view. That simply means: **with amount settings lower than 100%, what you see is not what you get anymore.**

Additional Displays

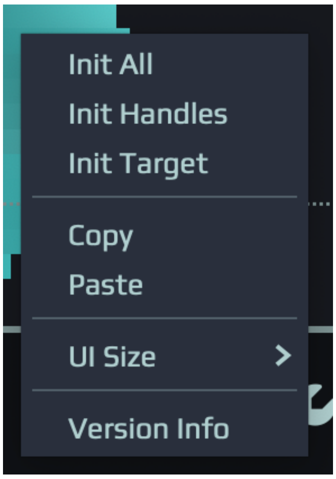
Below the main display, there are two buttons that toggle handy additional displays that overlay on the main histogram display.



With the **level meters**, you can see the peak and RMS levels of the input and output signals, just like the ones you'll typically find in your DAW. That way you can see how the dynamic histograms relate to these more traditional metering techniques.

The **curve display** shows you how the adjustments you make to the handles affect the Dynamic Grading's internal gain curve. If you've grown up with compressors, this is another way for you to connect with the "old world".

The Tool Menu



Clicking the wrench icon opens the tool menu, which is home to several useful features.

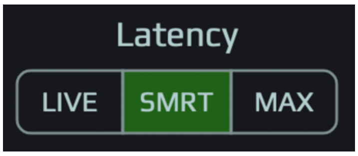
With the **Init Functions**, you can initialize different sets of parameters to their default values. "Init All" initializes all parameters, "Init Handles" initializes only all the graphical handles while leaving all other parameters intact. Finally "Init Target" also keeps the source handles intact and only resets the target handles to a neutral setting.

With the **Copy & Paste** functions you can transfer settings between plug-in instances using the system clipboard. The settings are encoded as text, so you can also – for example – drop a setting into an email and send it to a friend, or our support.

You can also set the **UI Size** to your liking and display a **Version Info** box with useful information about the version and host system you are currently running.

Dynamic Grading performs a daily **check for available updates**. If an update is available, a small update symbol appears on the wrench symbol and a “Download Update” option becomes available in the tool menu.

Latency Options



Like many advanced DSP algorithms, Dynamic Grading requires some **latency** for correct operation. Here, a delay is introduced to the signal to correctly align the measured loudness and the processed signal in time. **The latency must be equal to or larger than the response time parameter.**

When plug-ins report their latency to your host DAW, it can compensate for the latency so that all tracks are correctly aligned in time. When playing back your song, you will thus not notice the latency. However, changing the latency while the sequencer is running can lead to very annoying, unpleasant, and confusing effects in most DAWs, like tracks going out of sync, which will mercilessly interrupt your creative flow.

Starting with v1.2, Dynamic Grading offers three different latency modes: Live, Smart and Maximum. This is to ensure the best possible latency behavior depending on situation and task, without interrupting the creative flow.

Live Mode

In case of playing an instrument live through Dynamic Grading, latency cannot be compensated for due to causality constraints of the space-time continuum. In this case, you can **activate Live Mode to get zero-delay processing**. However, note that this will lead to the dynamic processing always being a bit “late”, depending on the chosen Response time.

But you can also **use Live Mode deliberately** as an artistic effect. When activated, the Response time parameter will act in a similar way to the attack time of a compressor.

Smart Latency

The new **Smart Latency** feature tries to **optimize latency** depending on the response time. This avoids situations where latency is lower than required or much higher than necessary. Also, in this mode Dynamic Grading tries to change its latency only when it's safe to do so, for example when the sequencer is stopped.

The current latency state is shown by the Smart Latency button's color:

- **Green: Latency is optimal.**
- **Yellow: Latency is higher than needed.** It's not optimal, but the audible results are "correct".
- **Red: Latency is lower than required.** As a result, audible results may be slightly incorrect.

When the latency is not optimal (the button is yellow or red), you should trigger latency optimization by stopping and restarting the sequencer. When doing that, the button will turn green again. You can also manually trigger latency optimization by clicking the Smart Latency button again, which may result in temporary track synchronization problems.

Maximum Latency

This used to be the normal operating mode in older versions of Dynamic Grading. It uses the worst-case maximum latency and is thus always high enough, but often much higher than necessary.

The maximum latency mode is useful when you are looking for the right response time setting, as it ensures that no matter how you set the response time, what you hear is always "correct". When you've found a good response setting, you should go back to Smart Latency mode again to reduce latency.

Further Information

Tutorials and Articles

For additional background information, tutorials and updates related to Dynamic Grading, please see our blog and the FAQs on our website.

Getting Help

For questions regarding Dynamic Grading please use our contact form or email us directly at support@playfair-audio.com.

Legal

Please go to our website to review our End-User License Agreement and Privacy Policy.