

Dynamic Grading



v2.0
User Manual

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About Dynamic Grading 2

Dynamic Grading is a powerful and versatile broadband dynamics processor. It has been designed from first principles in terms of what “dynamics” in audio really means, how humans perceive it, and how to analyze and manipulate it effectively and tailored to real-world use cases.

The heart of Dynamic Grading is the display of dynamic histograms, which provide a real-time visualization of the dynamic range. Paired with a versatile dynamic processing engine and intuitive graphical control, Dynamic Grading is a powerful workhorse for mixing, mastering and post-production engineers. Every day, and on every track.

This major upgrade is an evolution of the original concept with a completely redesigned user interface. Based on lots of valuable feedback from our customers, we made it easier to use for both new users and veterans.

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.

Overview



Dynamic Grading's user interface is built around the main display, which contains the **dynamic histograms for the input and output** signals and **markers to adjust grading settings**. In the area between the two graphs you can alternatively use the **ratio and gain knobs** to make the same adjustments.

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 and resetting the histogram visualization** and accessing other options like **source learn** and the **latency** options.

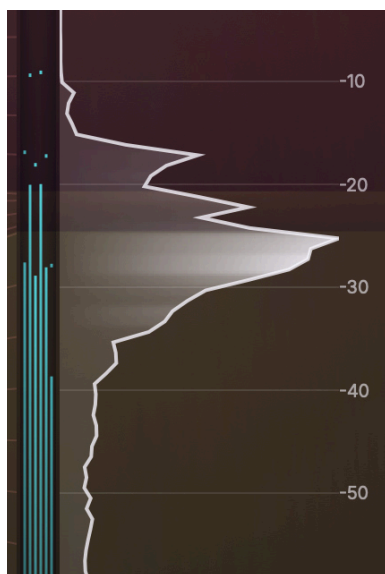
At the very top you have the on/off button as well as buttons for **Undo/Redo**, **Reference Compare**, and a **menu**.

In the sidebar to the right, you find controls that affect the measurement and processing algorithms. With these you can **fine tune processing** in a meaningful and musical way. The curve display shows how the current control handle settings translate to a gain curve.

Dynamic Grading's UI is **resizable** by dragging the corner at the bottom right. Double clicking the corner will return the UI to its default size.

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.



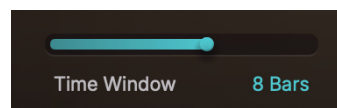
Imagine a really fast-moving level meter like the one on the left, and a superhuman statistician sitting in front of it, looking up the level on the meter really often and keeping a tally sheet of how often she or he reads each possible value over – say – the last 20 seconds. The result is plotted as a graph like the one on the right. In the example above, you can see that values around -25 dB have occurred much more often than values around -40 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).

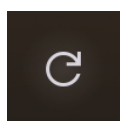
*A **sharp peak** in the graph hints at a static tone, note or noise with a **constant level**, while a **wide and flat shape** means there is a lot of **dynamic variation**.*

You can adjust the length of time considered for 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 **“Reset” button**. While changing relevant parameters, histogram computation is automatically accelerated so the changes made become visible faster.

Additionally to the long-term histogram, the brightness of the areas in the graphs show the **short-term loudness**.

*By **listening and observing** when which parts of the graph, 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

Dynamic Histograms provide a great overview of a signal's dynamic range. But the real power of Dynamic Grading comes from the ability to **shape the dynamic range in a creative and musical way**.

The Dynamic Process consists of these three steps:

1. Determine the input signal's **Punch, Body and Floor** ranges
2. Apply **compression or expansion** to each range
3. **Fine tune** parameters

We'll discuss these steps in detail in the following.

The Anatomy of Audio Dynamics

When looking at dynamic histograms, you will notice that most signals will show up more or less as a **bell shape**. Most of the time there will be a part of the dynamic range where most of the signal is concentrated, with the histogram **gradually tapering off** towards higher and lower levels.

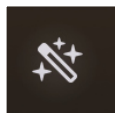
We call this the **Body Range**. It is where a signal carries most of its harmonically relevant contents, like the **sustain of notes and chords**, but also the medium-level hits in a percussion track.

Typically, some elements stand out of this Body Range. For example **accents, percussive attacks**, or particularly pronounced notes. These types of sounds are home to the **Punch Range**, which we call everything that is louder than the Body Range.

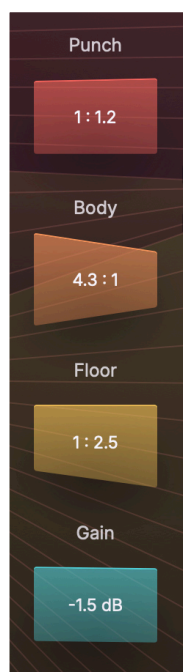
Similarly, there is also a region below the Body Range, which we call the **Floor Range**. This one contains the low level details of a recording, like **reverberation and echo**, but also unwanted elements such as **background noise**.



In Dynamic Grading, these ranges are determined by dragging the **source markers** on the left side of the main view. Everything between these markers is the Body Range. Everything above is the Punch, everything below the Floor.



You can also **determine an initial setting** for these ranges **automatically** by using the **Source Learn** feature. Simply play back a representative portion of your track to let the dynamic histograms build up. Then click the Source Learn button. It will analyze the source histogram and come up with a reasonable estimate of where the Body Range might be, which you can later refine if needed.



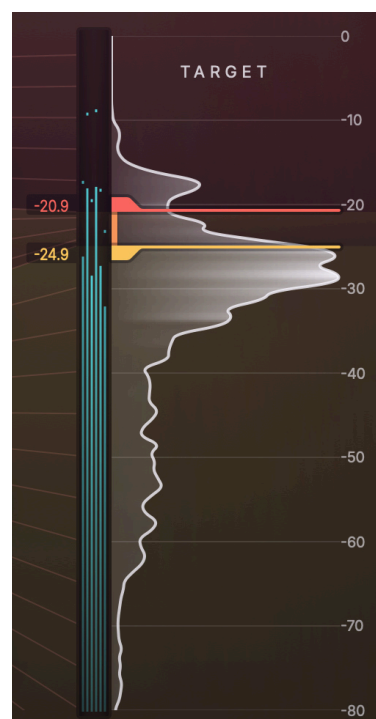
Shaping Dynamics for Fun and Profit

After determining the three ranges, it's time to bend them to our will. By **compressing or expanding each range individually**, we get a great deal of control over important aspects of audio tracks during mixing.

You can either control this compression/expansion using the **ratio and gain knobs** at the center of the plugin's UI, or by dragging the **target markers** on the right side with the mouse. With the marker lines, you **map the source signal's Body Range to where you want it to be in the output**. When hovering over them, dials will appear which control the **ratios for the Punch and Floor** ranges, respectively. The background graphics visualize what happens to the dynamic range on the way from input to output.

Since the Punch, Body and Floor ranges are home to different aspects of an audio track as discussed above, compression or expansion will have very different effects on the audible result.

In many cases, **starting with the body range is the ideal first step**, as it contains most of the relevant elements. By **compressing the Body Range**, you can for example reduce the difference between louder and softer notes. This will help **make the different elements remain audible even in a busy mix**. This is the primary task which classic compressors have been used for for



decades, but by applying compression only to the Body Range, some hard to control side effects on the signal's transients are avoided.

Conversely, by **expanding the Body Range**, the contrast between louder and softer notes can be increased. This can create **more breathing room and space** in a mix, making some of the softer elements demand less attention.

After initial adjustment of the Body Range, it's time to get to work on the Punch Range. Although we're applying the same kinds of processing - compression or expansion - the audible effects are dramatically different, because we are dealing with different elements of the audio.

In the **Punch Range**, we get **excellent control over depth perception** by modifying the transients and attacks. **Compression in the Punch Range** will reduce transients and peaks, which **increases the perceived distance**, all the while the overall loudness perception - which is determined by the Body Range - remains largely the same.

By **expanding the Punch Range** on the other hand, a track will seem **closer to the listener**. This is a great way to create a hierarchy of different tracks by making some instruments stick out of the speakers while having other lurk around further away in the background.

The final step is to deal with the **Floor Range**. In most cases, you will rather expand than compress here. Especially when compressing the Body Range, everything in the Floor Range will be amplified. This includes late decay of notes, reverberation and previously hardly audible background noises. If this is unwanted, **expanding the Floor Range** is a good way to get those low-level details back to acceptable levels, or get rid of them completely. With this approach, it's also remarkably easy to create the notorious Gated Reverb effect on drums.

But Dynamic Grading doesn't stop you from **compressing the Floor Range** either. Doing this will **exaggerate low level details** and can lead to interesting pumping effects.

Below is a handy cheat sheet to summarize the 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	Notes, sustain, harmony	Increase presence in the mix	Create space, enhance “groove”
Floor	Reverberation, noise, echoes	Exaggerate reverb, artistic effects	Dereverberate, reduce noise, gate

Fine Tuning

After having arrived at a first set of compression/expansion settings for each range, you will likely want to **fine tune the source range markers**. Using Source Learn or adjusting manually by looking at the histogram will provide a good starting point, but with some processing going on, you can now **use your ears** to make the final judgements.

By holding the **shift key** when dragging the source markers, you can move those independently of the target markers. Similarly, holding the shift key while dragging target markers will snap the respective settings to a set of meaningful values, like the same level as the source markers, or values where a corresponding ratio becomes 1:1.

Additionally, the side bar to the right provides you with some more parameters to fine tune the result. See the next chapter for a detailed description.

Further Adjustments

The Spectrum Parameter

To correctly deal with audio dynamics in a musically meaningful way, a dynamic processor such as Dynamic Grading needs to **track momentary loudness** more or less **consistently with human perception**. The sad truth is that almost all common dynamic tools such as compressors fail at this job.

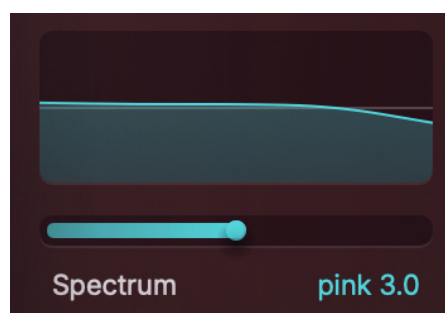
An essential factor in loudness perception is that of **spectral balance**, both in the way the human ear is constructed, but also what actual real-life signals are like. Nature has done a great job at matching both, while the electronics in most compressors and their digital counterparts still have to catch up.

Most real-life music signals (and many natural non-music signals) approximate a so-called **pink spectrum**, where the energy towards higher frequencies decays at 3 dB per octave, sometimes a bit more or less.

Human hearing is perfectly adapted to this by increasing its “measurement bandwidth” towards higher frequencies in the same way. However, the level tracking circuit in a compressor doesn’t do this. It assumes it is dealing with a white spectrum which doesn’t decay.

The Spectrum parameter in Dynamic Grading **compensates for the spectral balance** so the level tracking can properly deal with it. In its default setting, it will turn a pink spectrum into a white one before proceeding with level tracking.

*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**.*



To aid the Spectrum adjustment, the **Magic Eye** display just above the Spectrum slider **visualizes the spectral balance of what the level tracking sees**. Ideally, the curve should appear more or less flat.

After applying some compression/expansion, you can **fine tune the Spectrum parameter by ear** to a value that sounds most natural and musical to you.

A good way to explore the effect of the Spectrum parameter is by heavily compressing the body range 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 almost all traditional compressors work in a way that corresponds to the full white spectrum setting, which is something all audio engineers should be sad about.

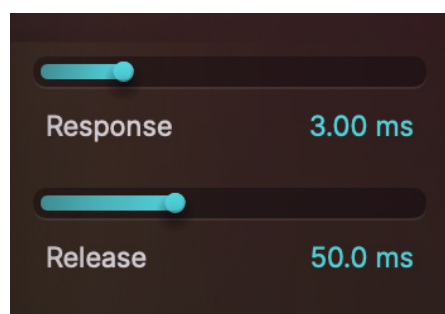
Timing Parameters

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

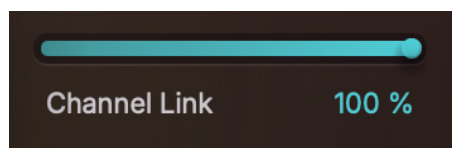
The **Response time** determines the time over which loudness is averaged for building the dynamic histograms and for dynamic processing.

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.



Variable Channel Link



The **Channel Link** parameter lets you seamlessly blend between two types of stereo or multichannel operation.

With **Channel Link at 100%**, all audio channels are **combined before analysis**, and the gain curve is applied according to that combined signal. This way **identical processing** is applied to all channels, **preserving the original spatial image** in the process.

With **Channel Link at 0%**, all channels are processed **individually**, as if they were going through separate instances of Dynamic Grading with identical settings. As a

result, the resulting dynamic processing applied to each channel may differ, which will typically slightly alter and **destabilize the spatial image**. This can be useful in order to add some more space and diffuseness to a signal.

The Curve Display

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 a way for you to connect with the "old world".



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).

Additional Functions

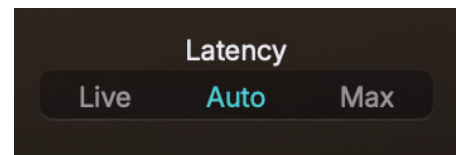
Latency Options

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 it so that all tracks are correctly aligned in time. When playing back your song, you will thus not notice any 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.

Dynamic Grading offers three different latency modes: Live, Auto 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 (we simply haven't invented time travel, yet). 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.

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 an ordinary compressor.

Auto Latency

The Auto 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 the first 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". Once you've found a good response setting, you should go back to Auto Latency mode again to reduce latency.



Undo / Redo

We assume that the usage of undo/redo buttons is self-explanatory.

Reference Compare

The **Reference Compare** Button provides an easy way to **compare your current settings with a previous "reference" setting**. By clicking the button, you can switch **back and forth** between the current and reference setting. The compare symbol is highlighted while you are listening to the reference.



By default, the reference setting is the one that was active when the plugin window was last opened. You can also reset the reference to the current setting by holding the Alt key while clicking the Reference Compare button.

Note: when you make new adjustments while the reference setting is active, the current setting that was active before switching to the reference setting are lost, as your new adjustments are now “current”.



The Menu

Clicking this icon opens the 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.

Further menu items include **resetting the resizable UI to a 100% zoom**, as well as the option to **disable the safety limiter**, and **set the Time Window parameter in seconds rather than bars**.

You can now also access, save or load **user and factory presets** in the menu. When you click save, the file chooser that opens will point you to Dynamic Grading’s user preset directory by default. Presets saved in this directory will automatically show up directly in the User Presets submenu.

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 open **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 menu icon and a “Download Update” option becomes available in the menu.

Multichannel Support

Dynamic Grading supports not only mono and stereo signals, but also a variety of **surround formats** to cover your dynamic processing needs across all three dimensions.

The **Channel Link feature works consistently** across all channel configurations, so you can seamlessly blend between joint multichannel processing and multi-mono processing of all channels individually.

The table to the right shows an overview of formats and supported speaker layouts.

Safety Limiter

The ability to expand the punch range upwards come with some risk of accidentally overdoing it. In such cases, an unexpectedly loud peaks at the input can become even louder when the punch range is expanded.

Format	AU	VST3	AAX
Mono	✓	✓	✓
Stereo	✓	✓	✓
LCR	✓	✓	
LCRS	✓	✓	✓
5.0	✓	✓	✓
5.1	✓	✓	✓
6.0	✓	✓	✓
6.1	✓	✓	✓
DTS 6.0	✓	✓	
DTS 6.1	✓	✓	
7.0	✓	✓	✓
7.1	✓	✓	✓
SDDS 7.0	✓	✓	✓
SDDS 7.1	✓	✓	✓
5.1.4	✓		
7.0.2			✓
7.1.2	✓	✓	✓
7.0.4			✓
7.1.4	✓	✓	✓
7.0.6			✓
7.1.6	✓	✓	✓
Quadrophonic	✓		
Pentagonal	✓		
Hexagonal	✓		
Octagonal	✓		

To protect your ears and equipment in such cases, while still allowing the occasional overshoot above 0dBFS (which you could address e.g. with a subsequent peak limiter), we have designed the **Safety Limiter**.

The Safety Limiter only becomes active when the **output peak level exceeds +6 dBFS**, and while active limits the output to a maximum peak level of 0 dBFS. After some time where no more peaks above +6 dBFS occur, it is automatically disabled again.

A red warning sign icon next to the “Output” label indicates when the Safety Limiter is active. You can disable the Safety Limiter at your own risk using the respective option in the menu.

Further Information

Website

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.