

Mixing with Guitar Pro 6: Panning, Equalization, Reverb and Dynamics Processing

Version 1.0
May 2010

Introduction	3
Panning	3
Equalization	4
<i>Decibels</i>	<i>5</i>
<i>Parametric Equalization</i>	<i>5</i>
<i>Gain</i>	<i>6</i>
<i>Shelving</i>	<i>6</i>
<i>Q factor</i>	<i>6</i>
<i>Example</i>	<i>6</i>
<i>Graphic equalization</i>	<i>7</i>
<i>Equalization hints</i>	<i>8</i>
Adding reverb	8
<i>Dry/Wet proportion</i>	<i>9</i>
<i>Time</i>	<i>9</i>
<i>Size</i>	<i>10</i>
<i>Colour</i>	<i>10</i>
<i>Pre-delay</i>	<i>10</i>
Dynamics processing	10
<i>Compressor</i>	<i>11</i>
<i>Expander Gate</i>	<i>12</i>
<i>Limiter</i>	<i>12</i>
<i>Gain</i>	<i>12</i>
Conclusion	12

Introduction

In Guitar Pro 6, you have a lot of effects processors, and a lot of control over them. The user interface usually presents them in a graphical way, similar to the way a real analog device would look, and the controls are named after their physical equivalent using the jargon of the sound engineering business. All those knobs and weird names may look daunting at first, so I'll give you a place to start by demystifying some of the most important elements: the equalizer(s), the reverb processor and the dynamics processor.

In Guitar Pro 6, each instrument track can currently have a chain of five DSP (Digital Signal Processors) attached to it, and you can define and store a total of four chains per instrument, so you can switch between these processing chains during the execution of the song by means of "automations". Say for example that you want the guitar to play clean all along, but you want to switch on distortion for the lead part. This you would do with a second DSP chain that would include a distortion pedal.

This works differently from Guitar Pro 5 where it was possible to alter various elements individually but in fact, the way Guitar Pro 6 does it makes more practical sense, because when one aspect of the signal processing chain changes, you usually need to alter other aspects too.

All the instrument tracks are then finally assembled and merged into the so-called stereo "master track", which can have its own processing, although somewhat different from the individual tracks.

Effects like distortion, amplifier choice, modulation etc. are certainly important to obtain the sound you are looking for, but having the ideal individual sound is completely useless if that instrument is swallowed in the final mix and does not stand out.

When mixing a performance or a song, the sound engineer (sometimes called "producer") will try to place each instrument so that it does not compete with other instruments for the same frequency band, and so that it occupies an assigned place in space.

In order to obtain a good final mix, the three most important controls are:

- the instrument panning
- the frequency equalization
- the reverb settings

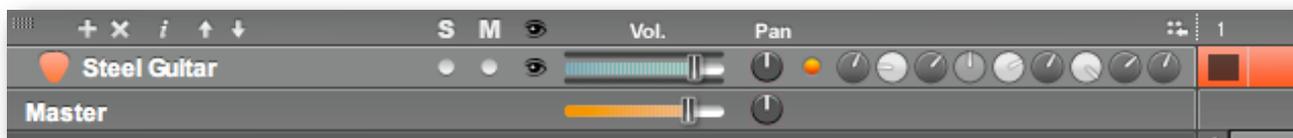
Dynamics processing is optional for mixing but we will discuss that as well.

Sound engineering is a well documented topic on the internet, with thousands of articles on the subject. This document certainly doesn't aim at being a reference on the subject; its only objective is to help newcomers get started with mixing tracks in Guitar Pro 6.

Panning

In order to obtain a good instrument mix, the most basic task consists in correctly "panning" the different instruments.

Panning controls the lateral placement of the instrument in the stereo mix (the so-called “panorama”). Imagine that you are looking at the band playing live. Close our eyes. You can clearly hear the lead guitarist somewhere on your right hand side (and moving!), while the rhythm guitarist is more on your left hand side (and maybe more static), and the bass maybe somewhere in the middle together with the drums. You can also clearly locate the keyboard, wind instruments, etc.



All this you can simulate with panning.

The figure above represents an instrument track in Guitar Pro 6. You pan an instrument by turning the Pan knob to the left or the right depending on where you want to place the instrument. This is a very sensitive knob, so you shouldn't have to turn it too much to hear a noticeable difference in instrument position.

The most important consideration when panning a song is to make sure that instruments with a similar frequency spectrum don't occupy the same physical place in the mix. For example, if the rhythm guitar playing power chords on a drop-tuned instrument is located in the same physical panning area as the bass, you are likely to saturate that part of the spectrum and create a rather muddy result. This is because both instruments produce the same range of (low) frequencies; as a result, the overall volume for those frequencies in that particular physical area is too high.

Panning is easy, but it should be your first action in creating a mix.

Equalization

Equalization is a process in which the sound engineer decides to alter the frequency characteristics of a channel by increasing (boosting) the gain (volume) of some frequencies and/or reducing (cutting) the gain of some other frequencies.

Every consumer Hi-Fi stereo amplifier offers a rudimentary three band equalizer in the form of tone knobs that let you control the amount of bass, medium and treble frequencies you want or don't want to hear. Guitar Pro 6 offers the same functionality, but with a lot more control.

In fact, Guitar pro 6 offers two different types of equalizers:

- a *parametric* equalizer available for each track
- a series of *graphic* equalizers that you can insert in the DSP chain

The difference between the two types of equalizers is as follows: in graphic equalizers, the frequency spectrum is divided into a number of bands whose width is fixed; in parametric equalizers, you can specify the width of the bands yourself.

In a graphic equalizer, you move cursors up (to boost) or down (to cut) frequency bands, and the resulting position of the various cursors is similar to the shape of the frequency response curve; hence the name.

In a parametric equalizer, the behavior of the device depends on the value you assign to the various parameters.

I recommend using the parametric equalizer for mixing, and a graphic equalizer for “special effects” in a DSP chain.

Decibels

In sound engineering, increases or reductions are usually measured in *decibel* (dB). This dB business can be confusing at times, because there is no single definition for a dB; all depends on what exactly is being measured.

In general, a dB is base 10 logarithm of a ratio:

$$\text{dB} = K \times \text{Log}_{10} Q/Q_r$$

where K is a constant, Q is a measured quantity and Q_r is a reference quantity.

If the quantities are voltages, the constant K is equal to 20; when the quantities are powers (volumes, pressures), the value of K is 10.

In the context of sound equalization, we want to boost or cut the volume of certain frequencies, so the second definition is more appropriate:

$$\text{dB} = 10 \times \text{Log}_{10} Q/Q_r$$

Hopefully you do remember some of your high school math?!

When the ratio is equal to 1 (which means that the measured quantity is equal to the reference quantity), the Log value is 0 (by definition of logarithms). So a value of 0 dB means neither boosting nor cutting.

When the ratio is equal to 2 (which means that the measured quantity is twice as big as the reference quantity), the Log value is 0.3 and the dB value is 3. So a value of 3 dB means a doubling of the gain.

When the ratio is equal to 1/2 or 0.5 (the measured quantity is twice as small as the reference quantity), the dB value is -3. So a value of -3 dB means dividing the gain by 2. In general, positive dB means boosting and negative dB means cutting.

Parametric Equalization

The parametric equalizer available for each individual track is a three band equalizer, very similar to the three tone buttons available on your Hi-Fi stereo amplifier. So basically, this equalizer divides the frequency spectrum of each instrument in three broad frequency bands: low, medium and high.

The difference, compared to the controls on your Hi-Fi stereo amplifier, is that here you have a lot of control over the definition of these frequency bands.



Let's have a closer look at the user interface of the EQ.

The leftmost orange button is an on-off switch for the EQ. When it displays orange, the EQ is active; if you then click on it it will turn gray to indicate that the EQ is bypassed (inactive).

In general, dark gray knobs in the EQ interface represent gain settings (amplification or reduction), whereas light gray knobs represent frequency settings. There is also one intermediate gray knob whose function will be explained later.

Starting from the far left, immediately right to orange on-off switch, we have:

- the low shelving gain (dark gray)
- the mid-3 frequency (light gray)
- the mid-3 gain (dark gray)
- the mid-2 Q-factor (intermediate gray)
- the mid-2 frequency (light gray)
- the mid-2 gain (dark gray)
- the mid-1 frequency (light gray)
- the mid-1 gain (dark gray)
- the high shelving gain (dark gray)

Finally, in the upper right corner of the interface there is clickable icon that lets you display or hide the equalizer (this is independent from the orange operational switch: the EQ can be simultaneously on and hidden).

In order to understand the function of the nine rotary controls of the equalizer, we need to delve into some more technical details.

Gain

This is the boost or cut applied to the frequency or range of frequencies, expressed in dB.

Shelving

In equalization, *shelving* usually means applying a constant boost or cut across a frequency range. If you would plot the gain versus frequency, the resulting curve would look like a flat shelf in the specified frequency range.

Q factor

The Q factor is indicative of the width of a frequency range. The higher the Q-factor, the narrower the frequency band, and vice versa. Another frequently used way to specify this information is by means of (a fraction of) an *octave*. An octave corresponds to a doubling of the frequency. For example, the interval between 440 Hz and 880 Hz is an octave. There is a mathematical relationship between the Q-factor of a bandwidth and its expression in fraction of octave:

$$Q = \frac{\sqrt{2^n}}{2^n - 1}$$

where n is the width in octave. So, a bandwidth of 2 octaves corresponds to a Q-factor of 0.66, whereas a bandwidth of 4 octaves corresponds to a Q-factor of 0.26

The Q-factor is used to indicate the range of frequencies around the center of the bandwidth that are boosted or cut.

Example

Let's look at a practical example. An acoustic guitar has a typical frequency range from 70 Hz to 1.2 KHz (including the harmonics). Depending on the musical context you might need to equalize the guitar in a variety of different ways. Here is one example:

Control	Value
Low Shelving Gain	+3 dB
Mid-3 Frequency	96 Hz
Mid-3 Gain	4.8 dB
Mid-2 Q	1,2
Mid-2 Frequency	4020 Hz
Mid-2 Gain	3.6 dB
Mid-1 Frequency	18 KHz
Mid-1 Gain	5,4 dB
High Shelving Gain	3 dB

The Mid-2 frequency is the center of the central band, and is set at 4020 Hz. The width of this band is very narrow with a Q-factor of 1.2. In other words, the Mid-2 boost of 3.6 dB is applied to a narrow range of frequencies in the middle of the spectrum. This is very typical when the guitar plays along with a singer, because the human voice occupies that part of the spectrum and you don't want the guitar to compete with it. When the guitar plays alone, you may want to give it a little more presence in the medium and widen the Q factor.

Around the low band frequency set at 96 Hz, we apply a boost of 4.8 dB. Please note that this boost is not applied *exactly* at 96 Hz (for reasons that we will not detail here), and also note that this boost is not applied *only* at that frequency. Instead, there is a range of frequencies around the set frequency that are more or less boosted. How much depends on the characteristics of the equalization filter (for real equalizers), but the slope on both sides of the set frequency is usually around 12 dB/decade (i.e. 12 dB every 10 Hz).

Around the high band frequency set at 18 KHz, we apply a rather large boost of 5.4 dB; apparently, this guitar was a little lazy in the treble!

Finally, everything below the low band or higher than the high band gets a boost of 3 dB (in this case the gain for the low and high shelving is the same).

The resulting setting is a moderately “scooped” EQ curve, with boost in the low and high frequencies and a smaller boost in the medium. This is very typical for guitars.

Graphic equalization

Guitar Pro 6 also offers a number of graphic equalizers, like the one displayed in this figure. They only differ in the number of bands and the way they divide up the frequency spectrum.

As you can see, this equalizer divides the frequency range in seven bands from 100 Hz to 6.4 KHz, and each band



center frequency is double the center frequency of the previous band (division in octaves).

You can adjust the cursors all the way up to +15 dB or -15 dB for each frequency band. You can also use the drop-down menu and select one of the preset settings available for that instrument.

Please note how this equalizer presents the typical scooped EQ curve for a guitar.

Equalization hints

The way you use equalization depends on how the various instruments stand in the mix. Say you have an acoustic guitar and you find that it sounds too boomy (a frequent issue with acoustic dreadnought size guitars such as the Martin whose samples are used by RSE); you can correct that by attenuating the lower frequencies for that guitar. Or say you find that the lead is competing with the synths in the higher end of the spectrum, and does not stand out well enough. Just reduce the higher frequencies for the synths so they won't stand in the way of the guitar. You can also use one of the proposed presets and see if that helps. This is all very subjective; you really have to listen to the overall result and decide by yourself. But beware of over-compensations that will sound unnatural. Make small adjustments at a time.

Like anything else in sound engineering, equalization is an art and not an exact science. Here are general recommendations:

- It is a lot easier to hear the effect when you boost rather than cut a frequency, but cutting is more frequently used when you try to improve the sound; boosting will often result in a "different" sound.
- Beware of over-boosting the low end of the spectrum; the result is usually very muddy.
- When boosting, it is recommended to use a large bandwidth (small Q-factor); for cutting, the opposite rule applies.
- Boosting highs will make the instrument stand out, while cutting highs will make it blend in.

Some people have tables with recommended settings for various instruments, but the best thing is to use your knowledge, reason and ear in each case.

Adding reverb

So far we have panned the instruments to place them laterally in the panorama, and we have compensated them in the frequency spectrum so every instrument neatly occupies its own frequency band.

But in order to create a realistic and pleasant sound mix, you also need to give it depth, i.e. place the drums and the bass behind the guitars (for example). This you control with the delay (or reverb) settings.

The physical principle is rather easy to understand. Physical sound waves leave the source and travel in all directions; on their way to you, the listener, they may be reflected any number of times on the walls and obstacles in the room. So what you hear is a mixture of direct sound waves and reflected sound waves; the more distant the source and the larger the room, the more reverb waves you'll hear. Reverb is essentially a much more

subtle form of echo, and is one of the ways our brain evaluate the distance of a sound source.

Digital reverb allows you to simulate all the above. In general, the original signal (called the “dry” signal) is split in two parts, reverb is applied to one part to obtain what is called a “wet” signal, and mixed back with the original signal.

Let’s have a look at how you can do this in Guitar Pro.

The figure represents the default reverb processor in Guitar Pro 6. There are other reverb processors available for insertion in a DSP chain, but we will ignore them.

On this processor, there are six knobs that you can use to finely control the reverb settings:

- Dry and Wet proportion
- Time (also called Reverb Time or RT60)
- Size
- Colour
- Predelay

Dry/Wet proportion

The Dry/Wet proportion is arguably the most important setting. It determines, in percent, the amount of direct waves (dry signal) and reverberated waves (wet signal) that reach the listener. In an anechoic room, the wet proportion will be close to zero (no reflections), and in a very reflexive room it will be much higher. But in mixing down a song, you will typically have a lower setting for the wet signal than for the dry signal, unless you have a lot of overdriven guitars, in which case you will need to increase the level of wet signal. This is because saturation tends to swallow reverb.

If you add up 100% wet to 100% dry, you will increase the signal strength and as a result you may get clipping (loosing some parts of the signal), so beware of that.

Time

The Time (or Reverb Time) is by definition the time (in msec) it takes before a reflected sound wave is attenuated by 60 dB (i.e. gets essentially inaudible). In reality, this value completely depends on the physical characteristics of the room and there is not much you can do about it in the physical world except using panels, carpets and curtains. But in digital sound engineering, you can change this time setting to very closely control (or “time-gate”) the reverb with respect to the tempo of the music. It allows you to have a rather large reverb (for example to simulate spaciousness), but tightly control the duration of that reverb and avoid that the whole mix gets muddy and weird.

Reverb time applies to all instruments, but it is absolutely crucial in getting drums (particularly snare) mixed down correctly.

In order to time a delay to land on the quarter beat, simply divide sixty thousand by the beats per minute of the song and you have, in a four four time signature, the number of milliseconds in every quarter beat. If you want an eighth beat delay, simply divide this



number by two, for a half beat delay multiply by two. For a sixteenth beat, a full beat (semibreve), just use a similar method.

Size

The Size knob will let you simulate the size of the room. Large room size settings for a particular instrument will give that instrument power and spaciousness, while small settings will keep things tight.

Colour

The Colour knob also has to do with the room. Smaller values for this setting will simulate a “warmer” reverb with more higher frequencies being absorbed, while higher values will simulate the opposite. This control actually simulates the different ways different materials reflect sound waves. You can of course also influence this with the frequency equalizer.

Pre-delay

Finally, the Predelay knob controls how far away the sound source is from the walls of the room. In fact, it measures the amount of time between the moment the direct sound reaches you and the moment the first sonic reflection reaches you. So, if that amount of time is very large, it means that the source is close to you and far from the walls since reflections reach you long after the direct waves. Conversely if that amount of time is small, it means that the reflected waves reach you almost simultaneously with the direct waves, and the source is closer to the walls and farther from you.

Dynamics processing

So far, all our settings have affected the individual instruments. But as indicated earlier, the individual tracks are eventually merged together in the so-called master track.

In audio engineering, “mastering” is a completely different business from “mixing”. Mastering is the process of putting the final mix on a master CD. But in Guitar Pro 6, “mastering” is simply a final adjustment you can do to the overall sound of the mix as a whole. These adjustments include:

- dynamics processing
- optional reverb
- optional equalization

These adjustments are meant to account for the differences between the mixing conditions and the listening conditions. You, the author and mixer of the tab, have mixed down the song in a particular sonic environment (say a spacious sleeping room) whereas I, the listener am listening to your song in a tiny office room. Also, our audio equipment is different.



This is the default user interface for the master track panel in Guitar Pro 6. As you can see, the reverb is completely optional, and in the 10 band EQ all cursors are flat at 0 dB.

The only element that we need to talk about is the dynamic processor in the top part of the interface.

When you listen to a record on your stereo, you will often decrease the volume when music plays too loudly, or increase the volume when you want to better hear it. Also, between two songs you will often reduce the volume to limit the rumble (noise) produced by the speakers.

All this is essentially what “dynamic processing” is all about, except that we will automate it instead of having to manually use the cursors for every action.

Dynamic processors fundamentally use two devices:

- a *compressor* to gradually reduce the volume as soon as it goes *beyond* a certain threshold
- a *limiter* is a special form of compressor that guarantees that a certain threshold will never be trespassed
- an *expander* to dramatically reduce the volume once it goes *below* a certain threshold

A compressor is typically used to compensate for the variations in volume of a performance; for example, a singer who goes from whispering to shouting, or an inconsistent drummer.

A limiter is often used to preserve monitors and speakers from dangerous excursions.

An expander is usually used to cut the rumble and noise coming from the speakers when the actual music stops.

Because a compressor reduces and levels the volume peaks, it effectively reduces the dynamic range of the performance (i.e. the difference between the max and min levels). Consequently, after compressing it is often possible to increase the overall gain.

Let’s have a closer look at the dynamic processor in Guitar Pro 6. It features all the elements mentioned above:

- An expander gate with control over threshold and ratio
- A limiter with control over threshold
- An overall gain knob
- A compressor with control over threshold, ratio, attack and release



Let’s now look at what all these knobs do.

Compressor

The *threshold* is the level above which the signal is reduced. The threshold is usually expressed in dB below peak; in other words, a value of -6 dB means that the compressor starts to act when the audio signal is 6 dB below its calibrated 0 dB mark. Very small values (e.g. -50 dB) mean that a large portion of the audio signal will be affected by the

compressor, whereas larger values (e.g. -5 dB or higher) mean the audio signal will be much less affected.

This is easy to understand, if you think it over. A value of -60 dB is a very small increase in volume, so the compressor will kick in very early.

The *ratio* is the amount of compression applied. It is usually expressed as a ratio of the form N:1. A ratio of 2:1 means that for every dB that the signal goes over the threshold value the signal is reduced by a factor 2. If the audio signal trespasses the threshold by 2 dB it will be reduced by 1 dB and leave the compressor 1 dB above the threshold. A ratio of 4:1 means that if the same signal trespasses the threshold by 2 dB it will be reduced so as to leave the compressor 0.5 dB above the threshold. So, the higher the ratio the stronger the compression. With ratios of 20:1 and more the compressor effectively acts like a limiter.

The *attack* time determines how long it takes before the unit changes the gain when a signal is applied. This may be very fast to prevent *anything* from exceeding the threshold, or deliberately slowed down to allow a percussive attack to the instrument. A relatively slow attack might be used with drums to increase the apparent dynamics, whilst actually reducing them.

The *release* time is the opposite; it determines how long the gain takes to return to "normal" after it has been reduced. A short release squashes the dynamics of the sound completely, and a very long release time ensures that all material remains at a constant fixed peak level, while still allowing normal variations. The overall dynamics (of a musical piece) are still compressed, since the soft passages will allow the unit to apply the maximum gain.

Expander Gate

As indicated above, the expander gate is used to reduce the volume when it drops below a certain threshold to control noise. The controls have the same meaning as for the compressor. You usually want a low threshold to prevent the gate to close too early and preserve as much of the original signal.

Limiter

The limiter only has a *threshold* control since its purpose is to prevent that threshold to be trespassed.

Gain

This is where you can increase the overall gain after applying compression. This is also often called "makeup gain".

Conclusion

This document has tried to introduce you to the most important elements of the Guitar Pro 6 user interface for mixing and DSP. There is of course much more to be said, but hopefully this will give you a place to start and will also give you enough background information to begin to make sense of it.