dominion
signal modelling device VST plugin

USER'S MANUAL

Date: 2002-08-24
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Subject: **dominion**, signal modelling device VST plugin, 'dominion.dll' (PC version), 'dominion' (MAC version)
Current program version: 1.2
MacOS version of this plugin compiled (and optimised) by Urs Heckmann (http://www.u-he.com)

*This manual describes the concepts behind **dominion**, its functions and the basic steps on how to use this software.*

***Dominion** is freeware and therefore free of charge. The latest program version is always available at the author's website.*

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What is dominion?

Dominion is called a *signal modelling device*. It can handle different jobs, all controllable by a simple and intuitive interface. Basically, it has 3 different sections:

1. envelope control. You can adjust the attack & sustain phase of a signal for fine tuning the transient response. Dominion's envelope control is most effective on percussive material, e.g. drums. The envelope control is pretty independent on the input level.

2. saturation section / nonlinear amplification. Static & dynamic saturation lets you create different tonal characters, ranging from digitally clean to warm sounds from the analog world. Dialling in 'dynamic' saturation produces the typical 'analog' timbre by compressing the peaks and introducing more (and different) harmonics on the transients.

3. high-frequency section. This works similar to an 'exciter' and creates additional harmonics that can greatly enhance the perception of clarity and depth of a signal. The extra harmonics are dependent on the saturation section in a way that 'hard' settings generate more and sharper hf details, while the 'soft' setting attempts to produce smoother mid & top end.

There is no need to use all available functions at once (okay, all you industrial people, go for it :D). It's you who decides how this tool will sound and how the stages are actually changing your source signal. Much is possible, from simulating vintage analog gear to drastically altering a waveform over time.

System requirements

Dominion is a real-time VST plugin. It is available for the PC/Windows platform as well as for MacOS. The requirements in particular:

PC/Windows:

You will need a PC machine equipped with Windows 9x, 2000 or XP, with reasonable speed for real-time audio applications. The minimum CPU power should be 200Mhz (one instance), more is better as the CPU power consumption increases the more instances you are willing to open.
A VST-compatible software host is required, such as Steinberg Cubase VST, Emagic Logic Audio, Orion from Sonic Syndicate/Synapse or hosts that are equipped with VST-to-DirectX adapters like Samplitude 6. Dominion has been tested with the above applications. There may be others which also work, but you will have to find out for yourself.

MacOS:

In order to run the Mac version of dominion, any modern G3 or G4 machine should work. As with the PC version, a higher CPU speed generally means a better performance and having more instances at hand.
Dominion has been tested with Emagic Logic under MacOS 9. If Problems with other hosts on this platform occur, please let me know.

Installation

That's pretty easy. As you are reading this manual, I assume that you have already extracted the archive (thereby using WinZip or a similar application on the PC or unstuffed it on the Mac). Its contents are

a) the plugin (PC: 'dominion.dll', Mac: 'dominion') and
b) the user's manual (yes, which you are currently reading).

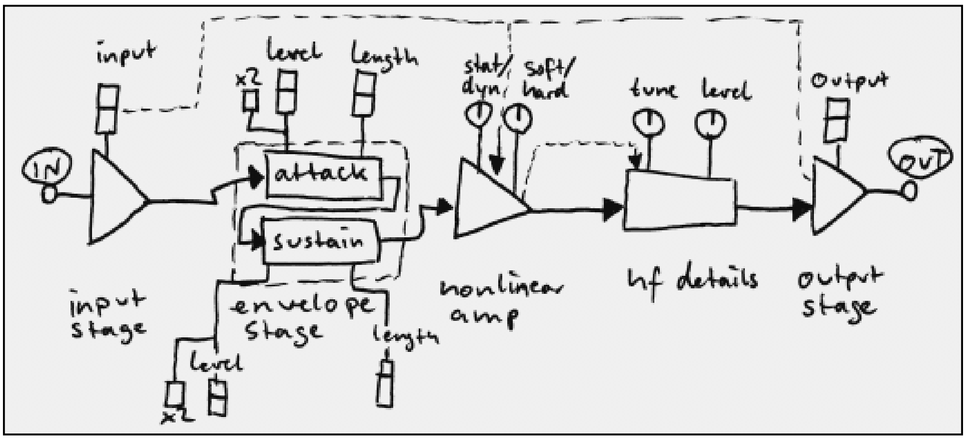
To install the plugin file, simply locate the folder named '\vstplugins' of your host program and copy the plugin file right into it. That's it. Now (re-)start the host. It will scan this folder and collect all plugins. When loading is completed, you should find dominion within the list of available 'insert' plugins.

What are 'inserts'? Erm, dominion is a dynamics tool, so it makes no sense to pass only a portion of the signal to it. Dynamic effects devices like this are made to process the entire signal to be as effective as possible.

This plugin comes with its own user interface. You should already be familiar with your host software so that you know how to open plugins and their interfaces/editors.

Control elements & their function

The following chart shows the signal flow within dominion:



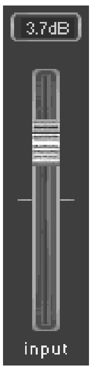
The different signal stages of dominion along with its controls will be introduced on the following pages.

Mono button

Pressing this button turns dominion into 'mono mode'. This means, a stereo signal is mixed up to a single monaural source. All further processing will only take place on one signal, so the amount of CPU power consumption is reduced.
If you are using dominion on a mono track anyway, you should let the plugin know it in order to save some calculation power for other purposes.

[Note: even when the 'mono' button is de-pressed for stereo operation, all the available controls are always affecting both channels simultaneously.]

Input stage



This is the very first stage of the unit. It is a linear gain device, providing with a range of +-6dB. If neccessary, you can crank up the slider to 'heat' up the source. Dominion won't introduce harsh digital distortion, but let your ears decide what is the right amount.

The input stage itself does not alter the sound on its own, but it controls the operation point of the nonlinear amp following the envelope stage. With increasing input level, the filter frequency of the (de-)emphasis circuit is shifted up and produces a slight roll-off within the treble region.
More on that circuit in the chapter about the saturation stage.

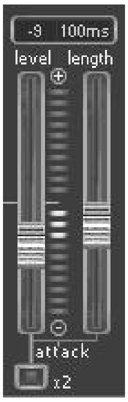
Envelope stage

This stage occupies 4 sliders and 2 buttons on the interface.
The attack & sustain level sliders have a center mark. Moving either one in the positive direction enhances the signal's attack / sustain portion. A movement in the negative region has the opposite effect: the signal will have less attack or less sustain.

The working principle is partly similar to a traditional dynamics compressor, but somehow it is not. Instead of reacting to changes in signal level, dominion makes a better guess as to what is actually going on (with a bit of help from your adjustment). This is due to what we call 'envelope'.

An envelope is simply the amount of volume a signal has over time. A typical compressor makes calculations on one of these (well, two for two-channel operation).
Dominion lets you control the 'attack' and 'sustain' part of the envelope independently. To do so, it uses 4 different envelopes, 2 for the attack, the other 2 for the sustain section.

Part I: attack section



To make things a bit clearer, consider the evelope of a snare drum: When a drummer hits the snare, there is an instant 'attack'. This noise is usually very short (only up to a few milliseconds) and (like most acoustic drums) of very high level.

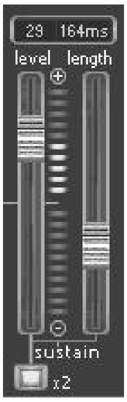
Now, what happens when you move up the 'attack' slider?
Dominion has built-in envelope followers. Simply speaking, those are 'riding the peaks' to estimate the momentary signal level. What they come up with is similar to what you see on a VU meter of a tape recorder: the needle does not just bounce up and down at every instant peak, it weighs the signal over some period of time, thereby approaching a level that we perceive as 'real volume' or loudness.
The attack section uses two envelopes: One to guess to signal's amplitude (as fast as possible) and a second one, which has a slower attack. How much slower depends on the 'length' setting.

So, what's left to do is to generate a control 'voltage' out of both envelopes' difference. By doing so, the only thing that could be different is the 'attack phase'. In other words: what you've dialed in with the 'length' slider is 'signal steepness'.
Usually, short setting will just emphasize the inital 'smack' of a signal (or reduce it at negative settings, thereby softening the waveform). Long settings boost or cut the entire attack phase. You should take care not to dial in settings that are too long as too much of the signal may be affected and therefore sound unnatural.

But the length adjustment should be long enough not to introduce waveform distortion which will be created in any system that does a large amount of volume changes within a very short period of time. Time constants that are too short are mostly audible at negative level settings. Your material might turn 'grainy' or 'rough' when it's overdone. Furthermore, it might sound dull because most of the natural perception of treble is due to transients. Treble information is a fast alternating waveform with a large steepness, but taking away transient response is like applying a low-pass filter which is actually nothing else but a function to even things out.

There is a button named 'x2' attached to the level slider. It doubles the available amount of attack enhancement or reduction.

Envelope story, part II: What's 'sustain'?



Continuing our example, when a snare drum decreases in level after the attack hit, you will only hear its resonance and that typical rattling sound (often plus the room in which the drum was played). This is sometimes referred as 'decay'.
Dominion is designed to be versatile (not only for drums!), that's why we are using the term 'sustain' instead.

You might guess that the process of calculating something that is supposed to be the 'sustain' is pretty simlilar to extracting the attack phase. Now, what the according 'length' slider does here is to determine the time it takes the second of the sustain envelopes to recover in comparison to the first which is the representation of 'natural' decay.

When you push the 'sustain' slider in the positive direction, dominion tries to 'hold' the tone for the time of the according 'length' setting by taking the signal's natural decay into account. The measurements will be taken over the period of time you've dialled in.

Often, the most dramatic results can be achieved at low 'length' settings as many signals have more useful information right after the attack rather than half a second later. However, raising the time constant might sound more 'relaxed' and less 'hyperactive'.

As with the attack, you can reduce the same amount of sustain by moving the slider into the negative region. Longer time settings will sound much smoother than short settings but may be less effective.

Like on the attack section, you can press the 'x2' button if the available level is not sufficient.

The attack & sustain control of dominion give you a wide and precise control of your source signal, whatever it may be. Applications are various.

As often with acoustic drums, a situation might occur when you have to process recordings where the individual drums are not perfectly dampened and produce a nasty booming sound after each hit or are prone to resonation. Dominion can help here simply by reducing their sustain phase. This works best on individual tracks by opening multiple instances of the plugin, but might also work good on sub mixes of the drum set.

Another example: guitars. Sometimes, the attack of a plucked clean electric guitar is too prominent: simply lower the attack level on the interface and make fine adjustments with the length. That's it.
Is it losing cripsness and definition in the mix? Well, just raise the level.
The beat is tight but the notes were played too long? Okay, trim them with the sustain slider.
The perfect solo, but the guitar had not enough sustain? Hm, I guess you know the story...

Generally, the less complex the signal is, the best performance you can expect from dominion. Sometimes, it might sound interesting on entire mixes, but this is playing with fire. It's always much easier to get useful information from less complex signals in order to make further calculations. Physics & statistics, not voodoo... :)

Now, try to answer the question 'can I achieve that envelope control with a traditional compressor'. The answer is 'no'. It's like comparing apples with pears. Dominion is no substitute for the typical tonal character of a compressor – it has a head of its own.

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Saturation stage (nonlinear amplification)

This is the tonal 'heart' of dominion. Following digitalfishphones plugin tradition, dominion uses methods similar to the ‘endorphin’ compressor to achieve an overall sound like that of analog circuits. As dominion is meant to be a creative device, you have plenty of control over the resulting sound the models give you. Furthermore, the saturation stage interacts with the other controls.

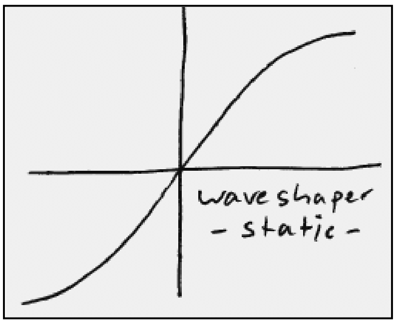
Like endorphin, the saturation uses a pre- & de-emphasis stage to achieve frequency-dependent saturation. As opposed to saturation the entire frequency band, this method greatly helps to keep the signal's clarity during high operating levels.

The pre-emphasis works by removing bass and lower mid frequencies prior to the saturation process. The de-emphasis stage takes care of reversing the filtering process to restore a flat frequency response. Well, this is not the whole truth, as the saturation process introduces new harmonics that add to the existing signal.
Dominion has a built-in roll-off at high frequencies that is more prominent with high levels in the input stage, similar to a hot-driven tape machine (btw, emphasis and generation of odd harmonics also apply for both devices).

You might say that dominion emulates vintage equipment and is capable of adding warmth. This is somehow true :)

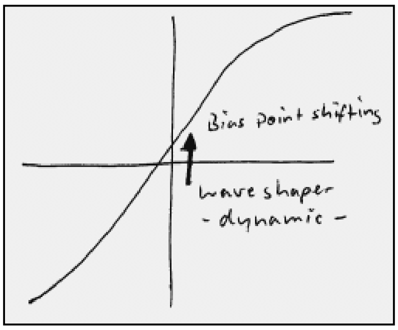
Harmonics structure

There is a knob ('static / dynamic') in the saturation section that controls the basic timbre of the saturation.



The harmonics are generated 'static' when this knob is turned down.

In this case, dominion generates only odd harmonics by applying a static so-called process of waveshaping. This is rather comparable to an analog device consisting of semiconductors, like transistorised record and replay circuits inside a tape recorder. Or just like any solid-state gain device.



Turning the knob (my ‘internal’ name is ‘style’, but it looked too flashy to me…) clockwise increases the amount of potential dynamic saturation. Here, some magic comes into play: the device generates odd & even harmonics on the transients and introduces a slight compression of the dynamics.

This working principle is very similar to what is going on inside tube-equipped gear: There is a detector that catches signal peaks and shifts the bias point for a short period. By doing so, the entire waveform gets distorted asymmetrically.

You should take a look on the level meter next to the knob: the more action there is, the more dynamics your signal might include. In other words, whenever the meter gets hyperactive, you can be sure dominion only distorts 'real' transients. Likewise, if the meter is standing still at some place, there is not much going on in a dynamic way, so it can only introduce odd & even harmonics in a static way. In order to achieve the typical warming 'tube' sound, you should adjust the 'input' slider along with the 'dynamic' knob most dynamic operation.

You are free to use the dynamic saturation feature for complex signals if your intention is just to warm things up and use dominion as a 'tube channel strip preamp'. The effect might be subtle but it's there and most certainly perceivable. Comparisons while pressing 'bypass' might be revealing at times.

The second knob of the saturation control is labeled 'shape'. In the anti-clockwise position, the most amount of saturation is introduced, the circuit's operation point is pretty low. This leads to a creation of harmonics from a very low level upwards. This setting is named 'soft' as the process has a smooth behaviour. The opposite is 'hard': Less amount of soft coloration will be perceived as only high-level signals will be affected. But when the stage gets bombarded with a hot input signals, the process will work more like a hard clipper that cuts off the top ends of the waveform. At times, this can sound less pleasent but is basically better the more complex the signal is as it ensures maximum transparency at typical levels.
You are right in saying a hard setting reveals less action of the dynamic process. This seems obvious because the transfer function is flat for a large portion of the signal, so there's not much going on in a non-linear manner. It always happens 'in the end'; should you hear some nasty distortion on the transients, try either lowering the input level, decrease the attack (or sustain if too much of a constant signal) or turn the shape knob towards the 'soft' setting.

High-frequency (hf) details

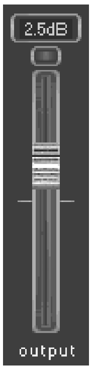
I guess you've heard about enhancers, exciters and similar units, along with all their myths. Some work by distortion, by equalisation, by changing phase response, some by magic... I
If we put all that marketing blahblah aside, nearly all of the different brands and units have all one thing in common: altering the overall sound in order to let things sound brighter or clearer, like as if you would take a veil off your speakers. Some people say it's true psychoacoustics and all about fooling the ear. Well, there are principles that apply to analog equipment anyway, for instance generating additional harmonics by distorting some portion of the signal. There are many reasons why tube amps sound pleasant to our ears. One of them is the fact that tubes are capable of producing large amounts of distortion, which – and this seems funny – doesn't sound harsh or nasty, but more 'rich' and 'balanced'. For ages, engineers have known that they just had to limit that distortion to mid and high frequencies. By mixing the signal with the original sound one gets the impression that things seem clearer.
A key to this perception seems to be phase response. So choosing the 'right' filter network is important and a way to get phase shift more prominent. Other manufaturers filter dynamically or combine equalisation with expander circuits. There are units that don't add any additional harmonics by not introducing distortion and relying on equalisation and delay (phase shift). In most cases, the effect is more subtle and best for signals that already sound good. It's the target sound and the application decides over the engineer's method.

Coming back to dominion: Some part of the audio is run through a high-pass filter. The filtered signal follows a saturation/distortion circuit to generate additional harmonics in a symmetrical way. This generation will be affected by the amount of 'dynamic saturation': sudden transients will produce more harmonics.

The 'tune' knob lets you control the frequency where 'it happens'; the harmonics are created above this point. By turning the 'level' knob clockwise the signal is enriched with the saturated hf signal.
Finding the right amount as well as the right frequency is largely dependent on the current application.
A basic setup could be to compensate for the high-frequency roll-off introduced by the saturation process at high input settings. Adding some shiny high end to the signal is common as well.

As with all those psychoacoustic devices, you need to be careful not to overdo the effect. Levels that are too high might sound pleasant in the beginning but may later be fatiguing to the ear. Setting the 'tune' knob too low can cause the ear to get irritated in the mid frequency region. As usual, use with care and compare it to the unprocessed signal from time to time.

Output stage



The output stage provides with up to 6dB of loudness to the whole signal, which means you can double the volume here. But you should listen closely. The extra gain is achieved by saturating the signal once more, but very drastic compared to the 'saturation stage'. Its main job is to ensure that the signal won't get digitally clipped. It's being kept at a level below -0.1dBFS, no matter how hard you push it. A symmetrical waveshaper takes care of the level, producing only odd harmonics.

This maximum operating level is high enough to use all of the available headroom while being compatible to older CD players (that might have problems with 0dBFS) and prevents you from cutting tracks the CD plant would certainly reject because of too much clipping with all bits set.
[Of course this is only the case if you insert dominion as a master effect.]

There is a LED at the output slider that lets you monitor the amount of saturation. It should not light up permanently.

While releasing the audio stream here, dominion needs to take care of clearing up the signal, because larger amounts of saturation always create a DC offset, where asymmetrical saturation introduces the most. In order to ensure a maximum headroom (limiting always relies on a symmetrical waveform), the signal will be DC coupled here by applying a 2nd-order high-pass filter at 30Hz.

Bypass button

By clicking the bypass button you will only hear the unprocessed audio just as if dominion was not loaded at all. However, your input signal is still being processed in its entirety – all control elements are still functioning. This lets you quickly switch between the 'dry' and processed signal state. I highly recommend that you make regular A/B comparisons to make sure you are still on the right track.

Reset button

The reset button instantly recalls the unit's default position for each of the control elements. This is the same as the ‘dominion init’ preset and is the easiest way to start from scratch.

A final note:

- Press the bypass switch from time to time, just to re-calibrate your ears.
- Don't be fooled by the term 'analog-style'. This is not the analog world. Never ever. Even if we're aiming at it with great effort, true analog circuits still sound different and keep a certain magic. This is not a story of good or bad, it is the question of 'What do I want to achieve? What is my way of working? What is the weakest thing in my chain?'. We shouldn't forget that.
- Give me some feedback on dominion, of whatever kind. In the end, it all helps to improve this software and influence my future developments.
- Have fun.

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Berlin, August 24, 2002
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