

# **FerricTDS**

**MANUAL**

revision 2.0

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# 1 Introduction

## 1.1. License

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## 1.2. Installation

Requirements:

- Win compatible system with SSE2 (or higher) instruction set support
- Tested and known to work in many VST compatible hosts

Put the DLL file contained in this archive in the VST plug-in folder of your host.

## 1.1. Overarching topics

**Warning: Lower your listening volume while operating the plug-in to avoid hearing damage or damage of speakers or any other equipment.**

Usage tips:

- Use the TRIM knob to level the outgoing audio and for handy A/B comparisons
- For optimum processing and results, adjust the incoming signal peaking to around 0 VU as shown by the VU meter in the middle
- The mkII version also allows for level calibration to bring the incoming signal onto duty level (plug-in internally)
- Use <ctrl> + mouse left click on a knob or switch to restore default position
- Use <shift> + mouse left click on a knob to fine adjust values
- Use this plug-in as an insert effect in any mono or stereo channel of your VST host

## 1.1. Credits

Visual concept by Patrick Barca, [www.suxesiv.ch](http://www.suxesiv.ch).

Many thanks to all the beta testers!

# 2 Jump Start

## 2.1. Overview

*FerricTDS* – Tape Dynamics Simulator.



Inspired by the smooth dynamic shaping capabilities of some high-end reel-to-reel tape recorders, this plug-in simulates three of the most distinctive and much appreciated sonic effects generated by these devices:

- gentle signal balancing and gluing it all together
- controlling peak performance
- harmonic excitation

Other side effects - such as pre-emphasis induced frequency changes, wow & flutter or noise - are not covered in this simulation.

## Functions at a glance

- performs gentle audio dynamic shaping
- balances difficult to handle audio material
- adds extra harmonic content
- saturates and controls outgoing audio peaks

## Plug-in specification

- Windows / VST compatible
- state-of-the-art digital signal processing
- performance-critical parts are written in assembler
- completely SSE optimized

## Getting the most out of it

Please read the following chapters to get the most out of this device and learn how to efficiently handle all parameters. This will help you to:

- understand the concept and workflow of this device
- understand gainstaging and calibration
- learn how to handle all parameters
- take advantage of internal routing

## 2.1. Quick reference

### Knobs and “screws”:

DYNAMICS – increases audio compression type effects

SATURATION – increases saturation effects such as harmonic content additions

LIMITER – controls the overall peak performance by limiting the dynamics output

RECOVERY – controls the recovery/release time of the dynamic processor

TRIM – adjusts the outgoing audio level in small ranges

SC OFF / 250Hz – high-pass filter control for the compressor's sidechain

CAL – calibrates the plugins operating level (0dB VU reference point)

### Switches:

ON / BYPASS – basic on/off operation

### Metering:

Main “VU” meter – indicates the the internal signal operating level after calibration has been applied

Horizontal meters – indicates the amount of signal processing in the DYNAMICS (left) and the SATURATION (right) processors by showing effective gain reduction amounts

## 2.2. New in version 2

With the mkII version of FerricTDS, further improvements have been introduced.

1. New plugin operating level calibration has been introduced which serves two purposes: better gainstaging at the outside of the plugin but also a completely output volume compensated processing. Additionally and to support this, the ballistics of all three meters has been carefully revised and aligned.
2. Now, an updated tape compression algorithm delivers punch and glue but with less IMD. While in its duty cycle, this compressor also creates additional 2<sup>nd</sup> order harmonic content on top.
3. Internal routing has been updated as well. To catch transients passing through the compressor, the limiter operates right afterwards now. Both together work in parallel to the saturation circuit.
4. The mkII version limiter is actually an oversampled ADC converter clipping simulation. This replaces the version 1.5 limiter which was based on envelope gain-riding.
5. All FerricTDS non-linearities are running at higher sampling frequencies internally.
6. A sophisticated analog signal path emulation has been added, providing some subtle qualities we all associate with the high-end analog gear.

## 2.3. Basic operation and advice

Use this plug-in as an insert effect in any stereo or mono channel of your VST host. It can be operated both as a mono or a stereo plug-in while both channels are handled separate internally (dual mono operation).

**Caution: If driven hard, this device can produce heavy distortion. Always lower speaker/headphone levels when inserting and operating this plugin!**

The best performance of this plugin can be obtained by leveling the audio input or calibrating the plugin so that sustaining signal levels do not exceed the 0 VU mark but peak levels occasionally do so. This indicates being in the “sweet spot” of this device.

Make sure that the ON/BYPASS switch is in ON position now. The two horizontal metering displays on the top left and right side are now responding to the incoming audio if processing actually occurs.

Always use a combination of both effects, dynamics and saturation. Dial in some DYNAMICS to obtain compression effects and SATURATION for some gentle distortion effects. Depending on the audio material, this effect can be rather subtle. Increase the input level to the plug-in or adjust the calibration setting if these effects appear too subtle.

If the compressor itself passes too much transient information, limiting can be utilized for compensation. Just increase the limiting threshold by turning the LIMITER knob clockwise to block outgoing audio signal peaks right after the compressor circuit.

All internal effects are output volume compensated but one can use the TRIM knob to adjust the overall output volume if needed. This feature is also handy for A/B testing at very precise equal volume levels.

## **2.4. Some tips on using the presets**

Explore the presets, but always adjust them to your current mixing situation:

- Always adjust the DYNAMICS and SATURATION parameters to your current track or mix in order to adapt the actual processing to your specific needs.
- All presets were designed on audio material with levels peaking at around 0 as indicated by the VU meter.
- Always calibrate the plugin and adjust the TRIM parameter for most accurate equal loudness levels during A/B comparisons.

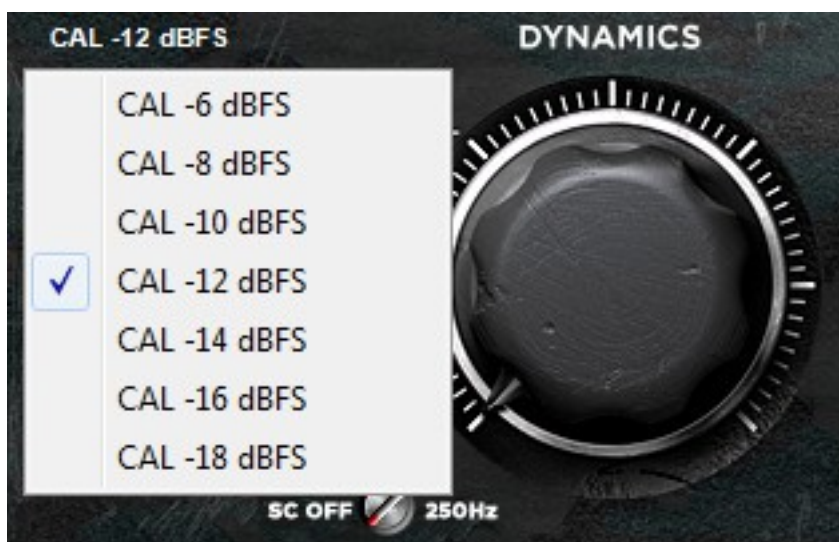


# 3 Advanced Usage

## 3.1. Gainstaging and Calibration

The CAL option offers a drop down menu to select one of seven different 0 dBu reference levels. FerricTDS offers -12dBFS by default which also reflects a typical analog tape calibration.

Using this calibration you can adapt any particular instance of the plugin to the required gainstaging in the host. On the other hand, this determines how hot the plugin operates internally. So this can also be utilized to lower or increase the internal circuits sensitivity.



The VU meter is calibrated accordingly and shows up proper levels within the plugin circuit sweet spot if sustaining signals stays below 0 VU and momentary peak levels are hitting the red zone.

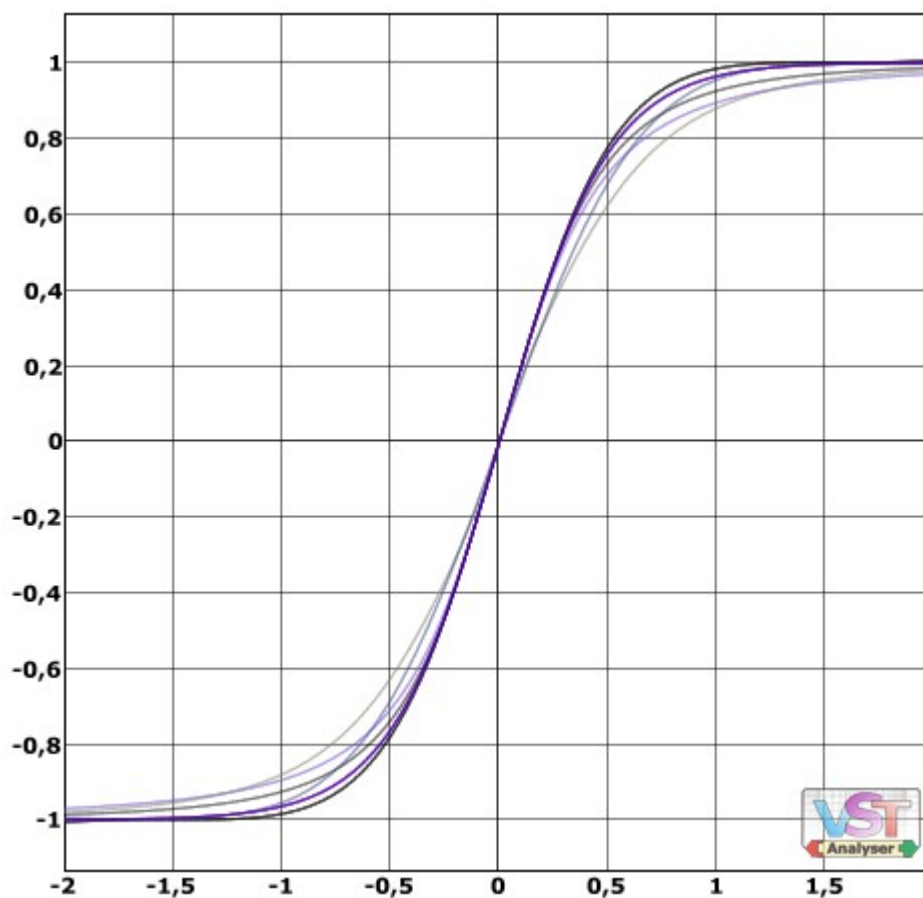
Staying within this sweet spot also assures equal volume compensated output levels over quite some gain ranges. But don't hesitate to use this feature creatively and push the device to its limits.

### 3.2. Internal Routing and Signal Limiting

FerricTDS internal signal path features a special tape compression and a tape style saturation unit in a parallel configuration. Both are mixed equally into the output stage and each path manages peak signal levels on its very own.

To avoid heavy signal overshots caused by transients leaving the compressor circuit, the compressor is directly followed by a limiter. With the mkII version the limiter resembles a special oversampled converter design (ADC) which can add impact and presence to signals as long as driven mildly. A clipping LED indicates if actual clipping occurs. **If driven hard, heavy distortion might take place. This is by design.**

To cut it short: In the wrong hands, it can be brutal :-)



Example gentle saturation curves

On the other side, the saturation circuit manages audio signal peaks on its very own given the nature of saturation. Within saturation, clipping does not take place abruptly but gets introduced smoothly.

Both signal paths are oversampled to manage aliasing artifacts.

### 3.3. Special Compressor Features

The compression unit of FerricTDS adapts to any program material on its very own. Just the overall amount must be dialed in with the big DYNAMICS knob (which can be seen as a drive/threshold control). However, the RECOVERY option allows to fine tune the release time behavior of the compressor envelope. This parameter lets you control how fast the DYNAMICS processor is recovering from its duty cycle.

Also, access to a highpass filter in the compressor sidechain path is given which allows to eliminate low frequencies from the compressors signal detector. It can be dialed in with the small screw ranging from 0 Hz (left-most / counter clockwise position) to 250 Hz (right-most / clockwise position).

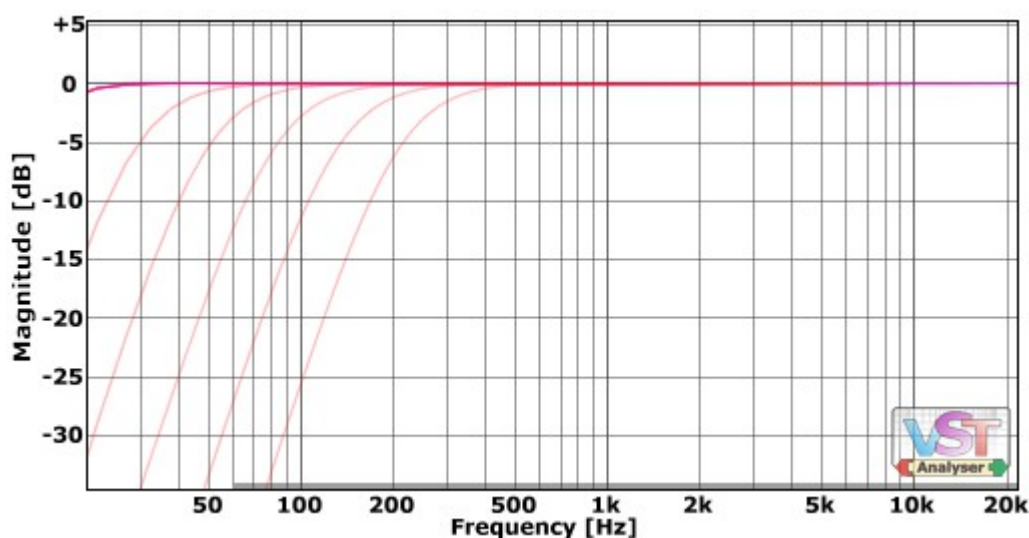


Effectively, this controls a 12dB/oct Butterworth high-pass filter in the sidechain path (**not** in the audio path itself) and operates from 0 Hz (aka OFF) up to 250 Hz. This filter lets you control the amount of low-frequency audio content that will affect the DYNAMICS processor. This parameter is also available to host automation (named “HP”).

This can be used, for example, to avoid “pumping” compression effects. Be aware that the more low-frequency content passes untouched the compressor, the more the clipper afterwards might be concerned. On the other hand, this might be a desir-

able effect, e.g. to apply more grit to a bass-line or bass drum by hitting the clipper stage.

When operating the plugin on a stereo track, the plugin actually performs dual mono processing. As with real tape processing, there is no stereo linking sort of thing. This is therefore by design.



Some example HP filter plots

### 3.4. Frequency and Phase Response

While internally taking advantage of parallel processing, the output signal does not have a 100% flat frequency and phase response (opposed to previous versions of this plugin).

This does not allow external dry/wet mixing of the processed effect signal.

# 4 Addendum

## 4.1. A brief history of tape

The concept of magnetic recording to a moving tape was invented by the German-Austrian engineer Fritz Pfleumer and received a patent back in 1928. The basic idea was to translate the voltage from the audio signal straight into magnetic energy, which then induces magnetic particles on a tape (moving along the inductor at constant speed). These particles manage to store the audio information. The whole process goes the other way around for recall.

Although this was a revolution for both broadcast and recording industry, there were many technical challenges to be addressed before its success during the middle of the last century. Some physical limitations can't be ignored even today. While electro-mechanical problems, such as wow and flutter or noise and crosstalk have been improved over the years, the electromagnetic phenomena, such as magnetic permeability, hysteresis or the Barkhausen effect still must be addressed.

Additionally, since a tape can't store unlimited amounts of energy, a natural saturation occurs when signal levels are driven too hot. Normally this has to be avoided, as it can lead to heavy distortion. Nonetheless, this type of saturation was (and still is) frequently used as an artistic audio effect.

The new digital recording technologies that emerged towards the end of the 20<sup>th</sup> century overcame these shortcomings of analog recording and made tape obsolete – if regarded from a purely technical and workflow-related point of view. Yet some of the positive effects of high quality tape and recorders are still highly appreciated in today's audio production, and there is quite a lot of myth and buzz going on about it's "magical" qualities.

In fact, what makes a good tape and recorder still attractive in the digital age is its overall ability to balance audio dynamics while adding harmonic content and gently limiting the peaks. If properly applied, this can result in a very pleasant sonic experience. However, it still comes at the expense of some of the mentioned artifacts and side effects, not to mention the time and cost of operation and maintenance.

#### **4.2. Judging saturation effects**

There are quite some mistakes floating around on how to judge a saturator's sonic quality and here are some tips to avoid the most common pitfalls:

1. A good saturator does not appear as distortion in the very first place. Firstly it just saturates incoming audio signals which means that at a similar RMS output level it simply reduces the peak performance (which results in a smaller "crest factor").
2. This immediately implies that you need a RMS meter in your output chain to compare different saturation settings or devices to another. Basically this is the same when comparing limiters or maximizers.
3. Distortion is a side-effect which typically occurs at higher saturation levels. It can have different sonic qualities, e. g. due to the frequency distribution of distortion which makes a huge difference to human hearing and if the effect is perceived as to be rather gentle or not.
4. Listen and pay attention to side-effects like spectral balance, perception of depth and dimension and how signal transients behave.
5. A good saturator can also glue complex audio together which might be a rather subtle but pleasant affair.
6. Don't rely here on a simple spectrum analyzer since it does not reflect/know anything about hearing perception.

#### **4.3. Updates and further information**

Refer to my Blog at <http://varietyofsound.wordpress.com> for some additional information and updates on this plug-in or leave a note there if any issues did occur.

Peace,  
Herbert