# A study on plug-in effects and DAW project sample rates.

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## Preface

Deciding on a S.R. (Sample rate) when starting up a new project in a DAW (Digital Audio Workstation) may sometimes be a pain. It is said that working in a higher S.R. results in a more clear sounding master. However, there is a price to pay in order to work at higher rates... Hard disk drives with higher read/write speeds to process larger recordings, a faster CPU to crunch increasing data, etc, etc.. There are also countless discussions on wither there is a point in working at higher S.R. when the final product (often CDs) only has a capacity of recording up to 44.1kHz, truncating everything above anyway.
This paper focuses on only one of the many aspects of editing in a DAW where the project S.R. may effect the resulting product. That is, when adding harmonics via software plug-ins. This includes saturators, compressors or any effect with potential to add harmonics to the original signal. Often, those are the type of effects said to “add warmth” and/or “fatten up the sound,” but often not sounding as good as the classic analog equivlent.

This paper does not discuss any listening experiments. How the signal's audibility is effected by the processors is often a subjective matter which varies depending on the listener's experience and/or listening environment. Instead, we will only view frequency distribution charts.
You are encouraged to reproduce the results on your own.

## Overview

We will compare frequency distribution charts of a single “sweep” processed in differing sample rate projects, then exported at 44.1kHz.
Just to clarify, the files we are comparing (exported audio file) are all 44.1kHz, which means their potentials as a container are equal only the process to create them differ.

**Preparation**

- Cakewalk Sonar (I used 8.5 Producer)
- Wavosaur (A free 2tr wave editor) http://www.wavosaur.com/

I will be using more of effects included in Sonar and freeware plug-ins, using only included preset settings for the ease of reproducing. As stated earlier, I encourage you to try the same with your favorite plug-ins. A similar experiment should be possible on DAWs other than Cakewalk as well.

**Procedure**

・*Procedure overview*

1) Generate an audio file containing a sweep tone and save as a 44.1kHz file.
2) Open the above file in multiple DAW projects, each project with a differing S.R. settings.
3) Insert the effect of your choice in each of the projects.
4) Export each project to a 44.1kHz file.

・*Procedure detail*

<u>In Wavosaur</u>

1) Create a new file (1ch/44100/32bit float/10sec.)
2) Create a sweep (Tooks->Synthesis->Frequency Seep->Linear)
3) Slightly fade out the end of the sweep（To protect your speakers）
4) Add 5 seconds of silence to end of the file （Extra headroom for effect tails）
5) Save the file.

<u>In Sonar (or any DAW of your choice)</u>

5) Create multiple projects, each with differing S.R. settings.
6) In each project, import the sweep file created earlier in Wavosaur.
   (In the case of using Sonar, the file will automatically be up-converted to the project S.R. when not 44.1kHz)
7) Insert an effect of your choice（Make sure the effect settings are equal among projects.）
8) Export the project to a “.wav” file(mono/44100/32bit float)

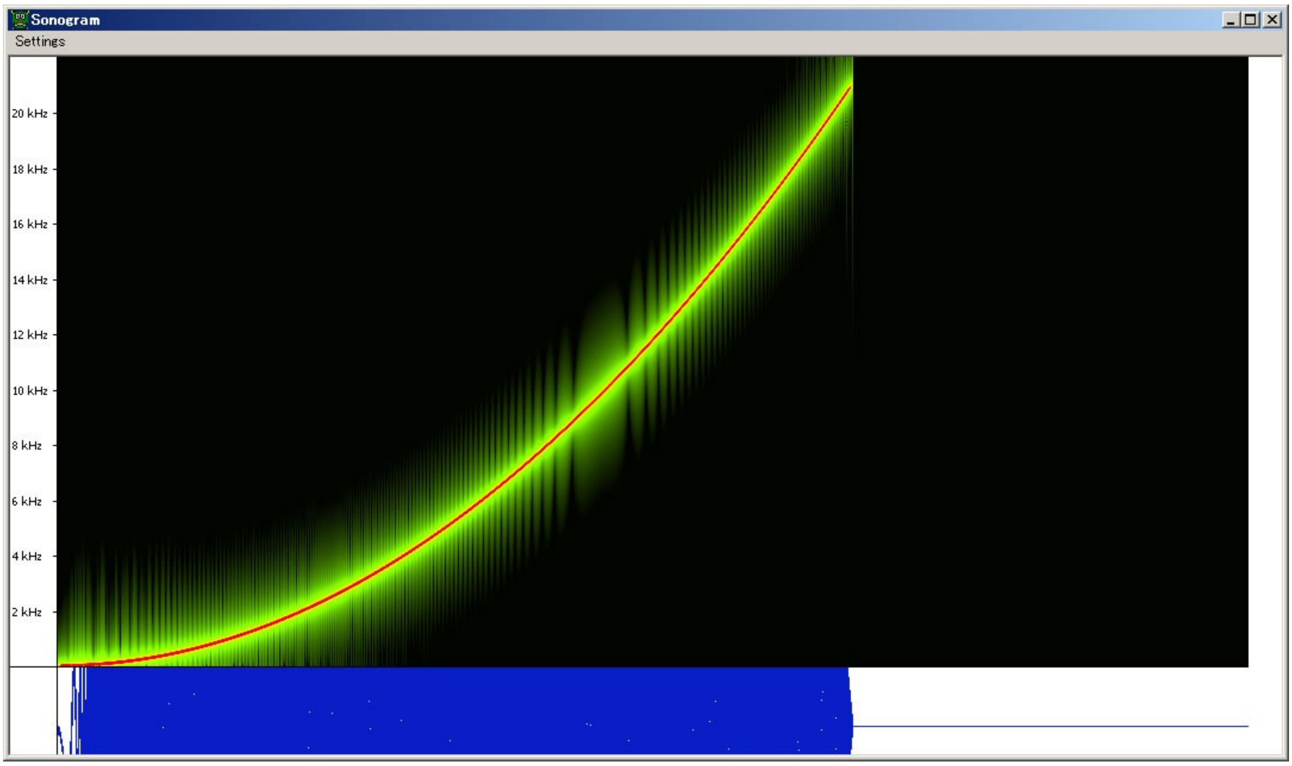
<u>In Wavosaur</u>

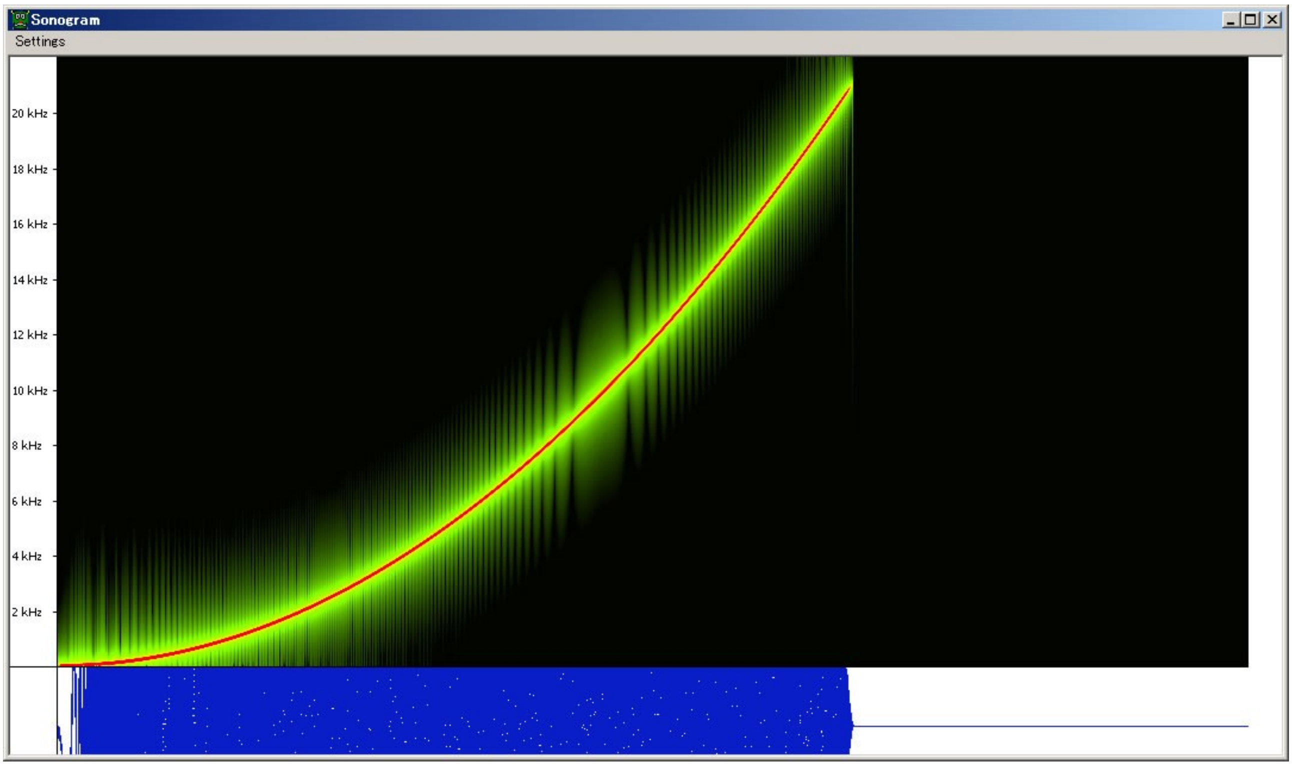
9) Open the files exported in Sonar.
10) Display the frequency distribution（Tools->Sonogram, or press “G” on your keyboard.)

## Results 1

Above: 44.1kHz sweep. This is how the source we will later alter looks before processed.
Below: The above file loaded in an 88kHz project, then exported to a 44.1kHz file.

Description:
The first two charts are to confirm that a simple up-conversion → down-conversion in the DAW causes no change in the signal as far as for visual recognition.

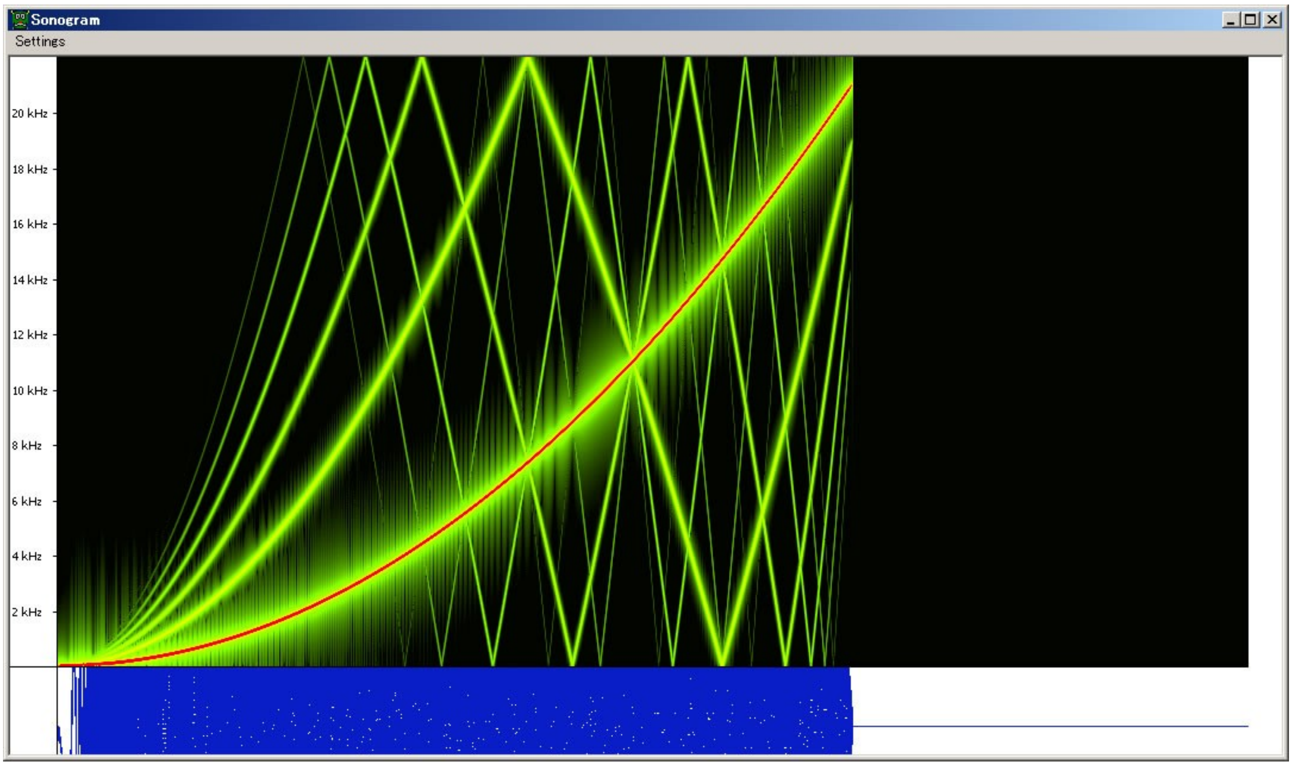


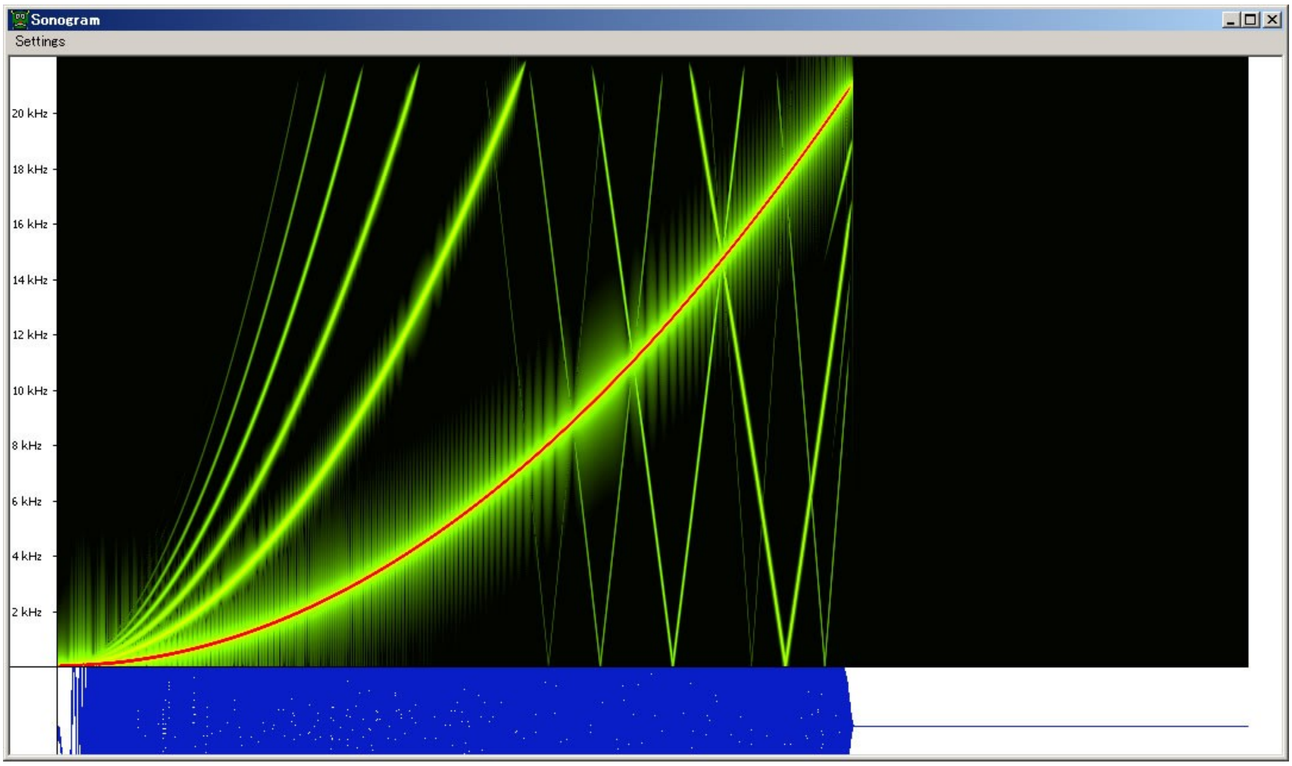



**Bootsy FerricTDS** （Preset “Classic Tape”)

From above, altered in 44.1kHz, 88.2kHz projects.

Description:
You can see from the charts that this effect preset is adding at least 5 harmonics above the original signal. You can also see in both charts; more obvious in the 44.1kHz project; that the signals are folding over once exceeding half of the project S.R.

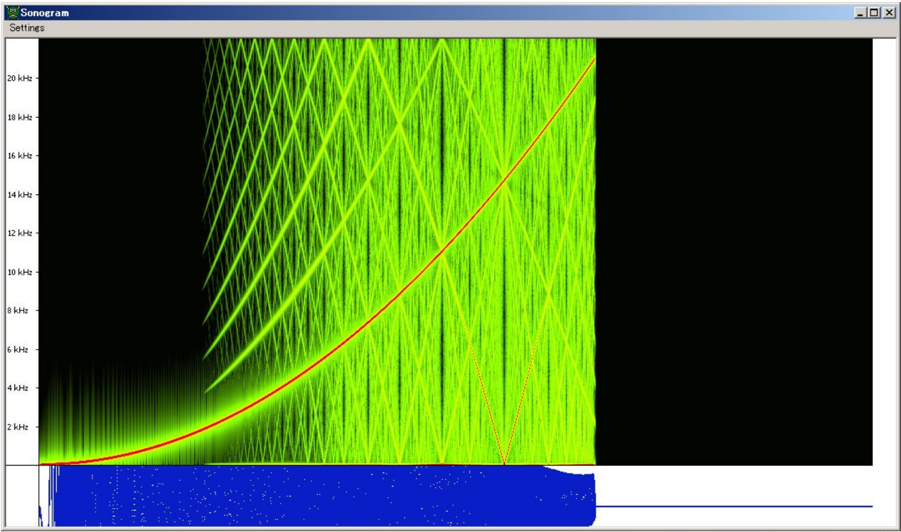


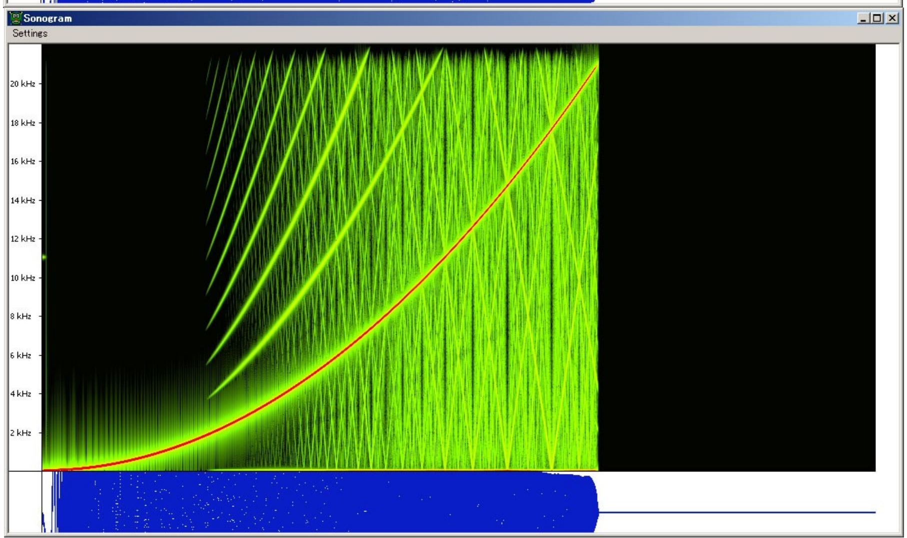



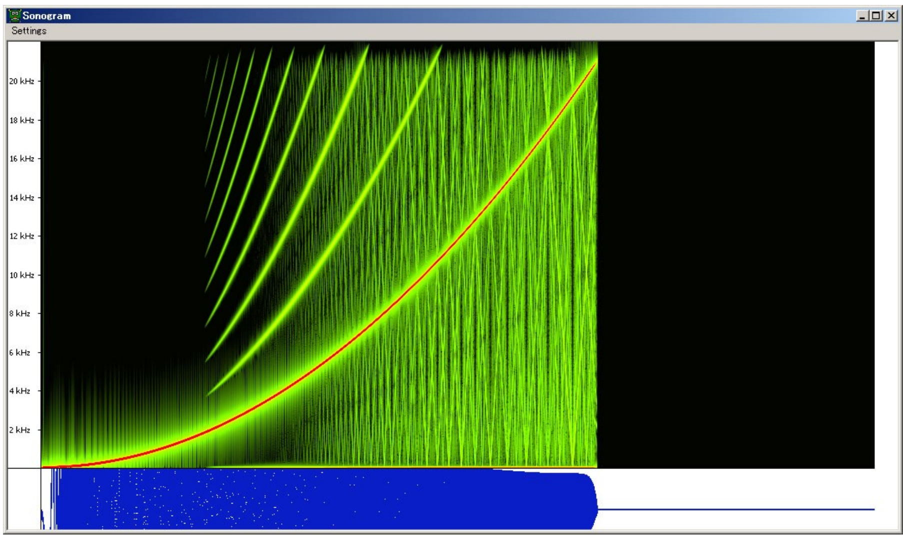
**Cakewalk Tube Leveler** (Preset “Mix Warmer”)

From above, altered in44.1kHz, 88.2kHz, 192kHz projects.

Description:
Perhaps by design of the effect, the signal is not altered with inputs below 2kHz, then for what is exceeded, a series of harmonics is added.
As seen in the previous case, the signal is “folding over” for frequencies above half the project S.R.



**Conclusion**

Most of the modern DAWs and digital audio formats handle audio in a method known as PCM. (Pulse Code Modulation) When an analog signal is recorded or altered in this domain, a phenomenon known as “aliasing” or “fold over” effects signals of frequencies above half the sampling rate. This phenomenon is explained in detail in most elementary guides to digital signal processing or digital audio. In short, when recording at 44.1kHz, a 33kHz signal will fold over and appear as a 11kHz signal (33 minus half the sampling rate)

What is going on is, your editor is not only making your signal less clean, but “forging” signals which did not exist, and yet is relevant to the input signal. These are also in a sense noninteger harmonics. The human ear is generally more sensitive to such harmonics other types of noise with a random characteristic.

In the case of working in a 192kHz project, you have a head room of 96kHz (192/2) to exceed before such artifacts are introduced. In addition, signals slightly exceeding 96kHz and folding over beneath are still way above the audible range. The system can then filter out signals above 22kHz before down-converting to 44.1kHz.

Just to remind you, the human ear is sensitive to frequencies around up to 20kHz. In theory, the CD format of 44.1kHz (records frequencies up to 22kHz) is enough to cover the audible range. However, an editing medium of 44.1kHz has no headroom for treating harmonics added to the source. Anything that accidentally  spills above 22kHz will instantly “forge” tones in the human audible range.

As a personal note, I stand against the common myth that a high S.R. is not necessary for mixing noisy genres such as rock which is already full of noise. This may be true if you are recording digitally but applying effects for loudness in the analog domain (i.e., using analog outboards) However, in case of concluding the mix inside a DAW (a.k.a. “mixing in-the-box”), working on sample rates close to the final master will result in forging noninteger harmonics which accumulate each time another saturator is inserted. The sum of such signals maybe what make DAW mixes sound “less warm” than when mixed in conventional studios.

Therefore, the more you intend to later insert digital effects which may generate harmonics, there is more of an advantage working at a higher S.R.

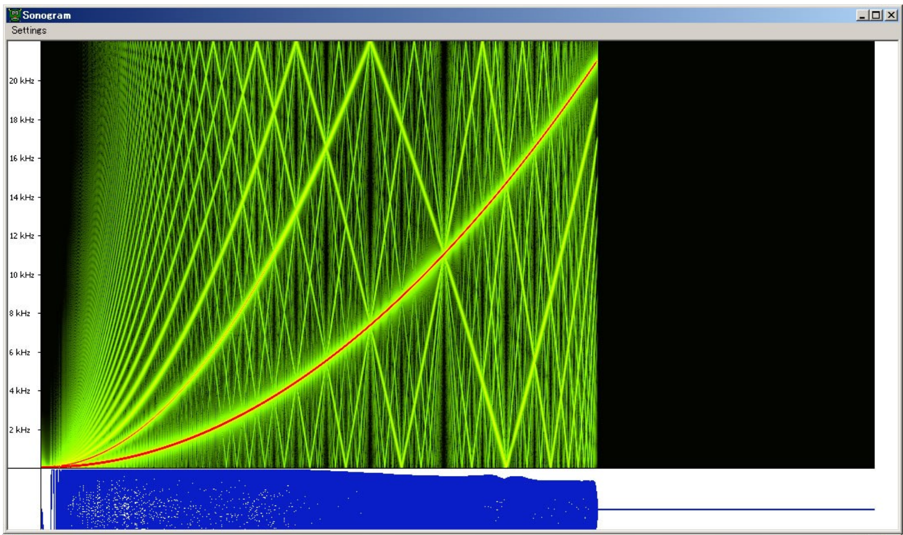
Just for the pleasure, following pages will illustrate some more examples.

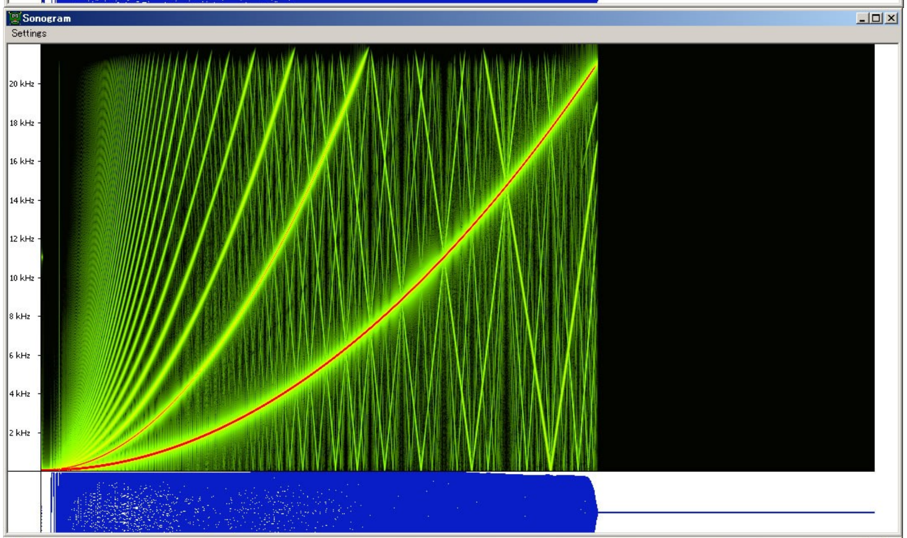
**Results 2**

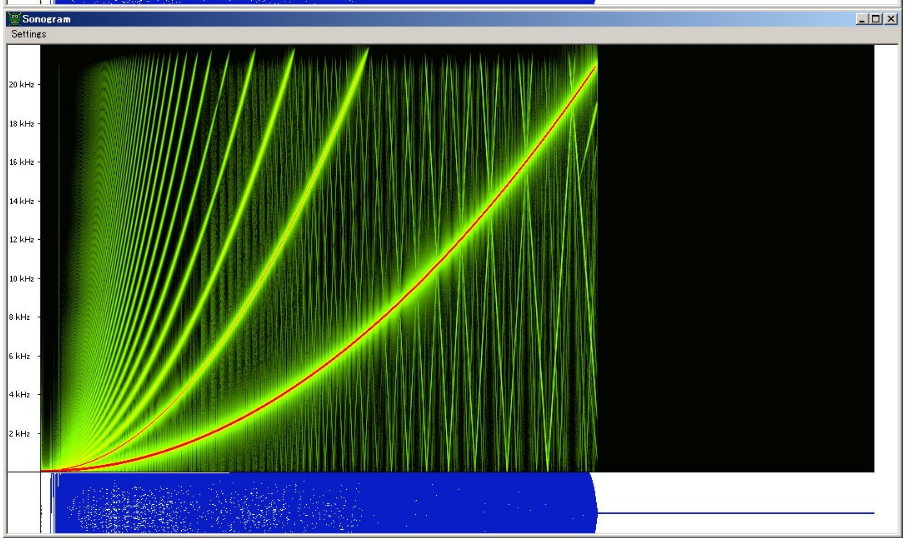
**Cakewalk VX64**(Preset “Heavy Distortion”)

From above, altered in 44.1kHz, 88.2kHz, 192kHz projects.

Description:
Unlike the previous Tube Leveler (also from Cakewalk) which affected signals above 2kHz, this plug-in adds harmonics to signals further down. There are fold overs in each project settings, but is reduced as the S.R. setting increases.



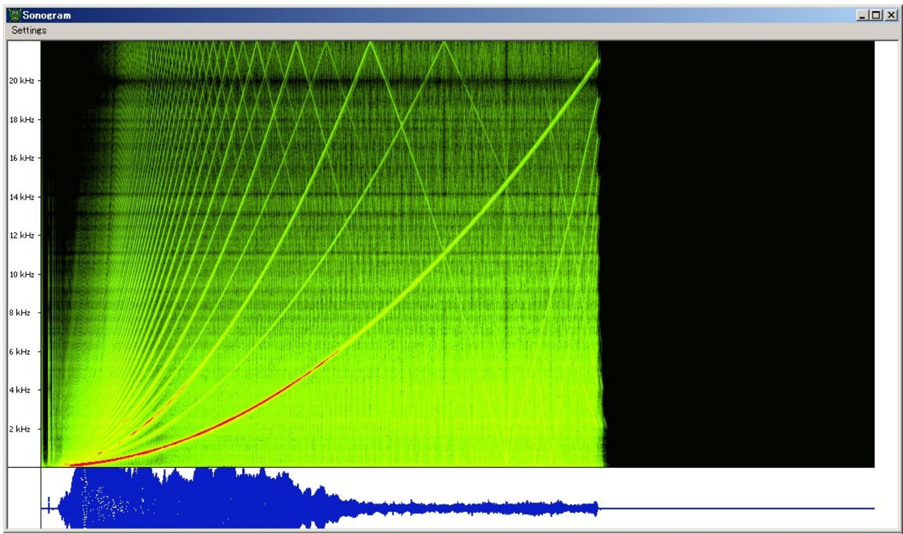



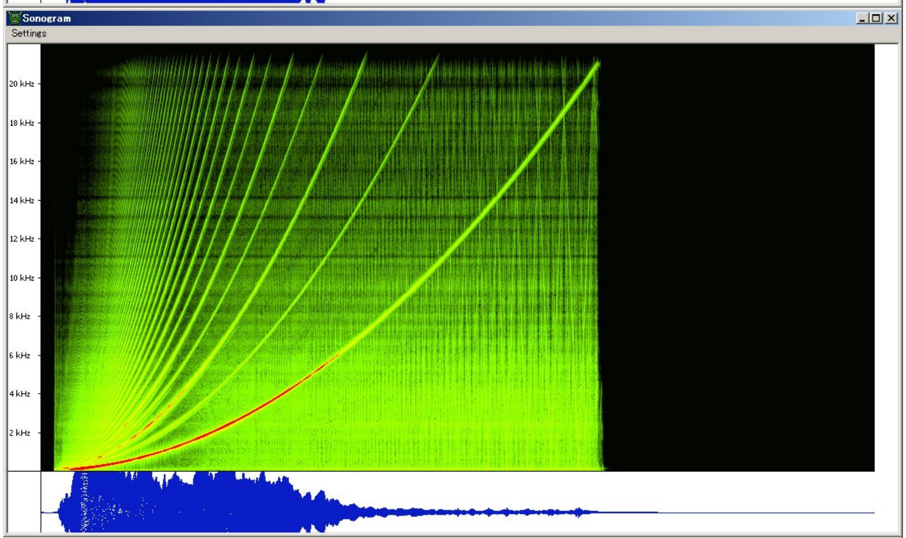



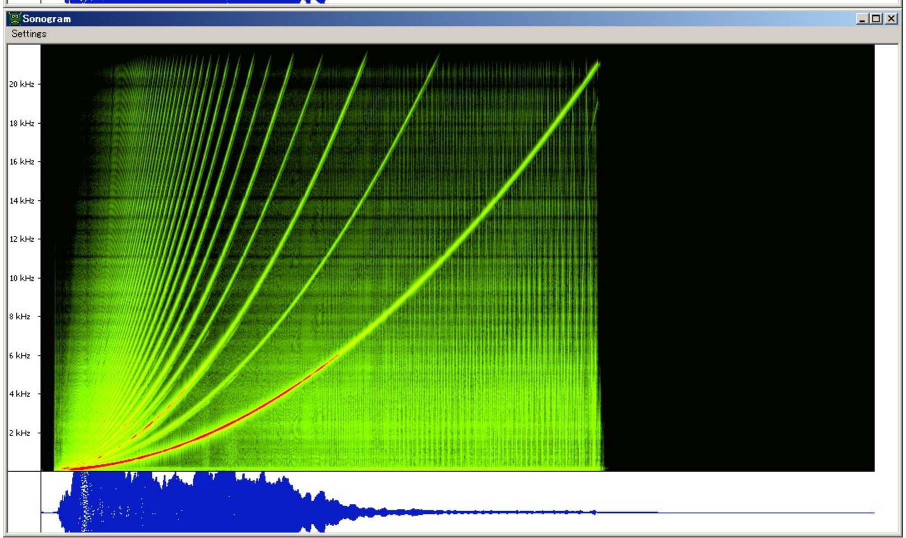
**Native Instruments Guitar Rig4** (Preset “Jimi's white pleasure”)

From above, altered at 44.1kHz, 88.2kHz, 192kHz projects.

Description:
An example of a distortion effect for guitar. Distortion is often thought of as a method of adding noise and noise masks other subtle signals. However, at lower S.R. you are forging noninteger harmonics which stand out more than noise suitable for masking. This may be where often appreciated “warmth” is lost.



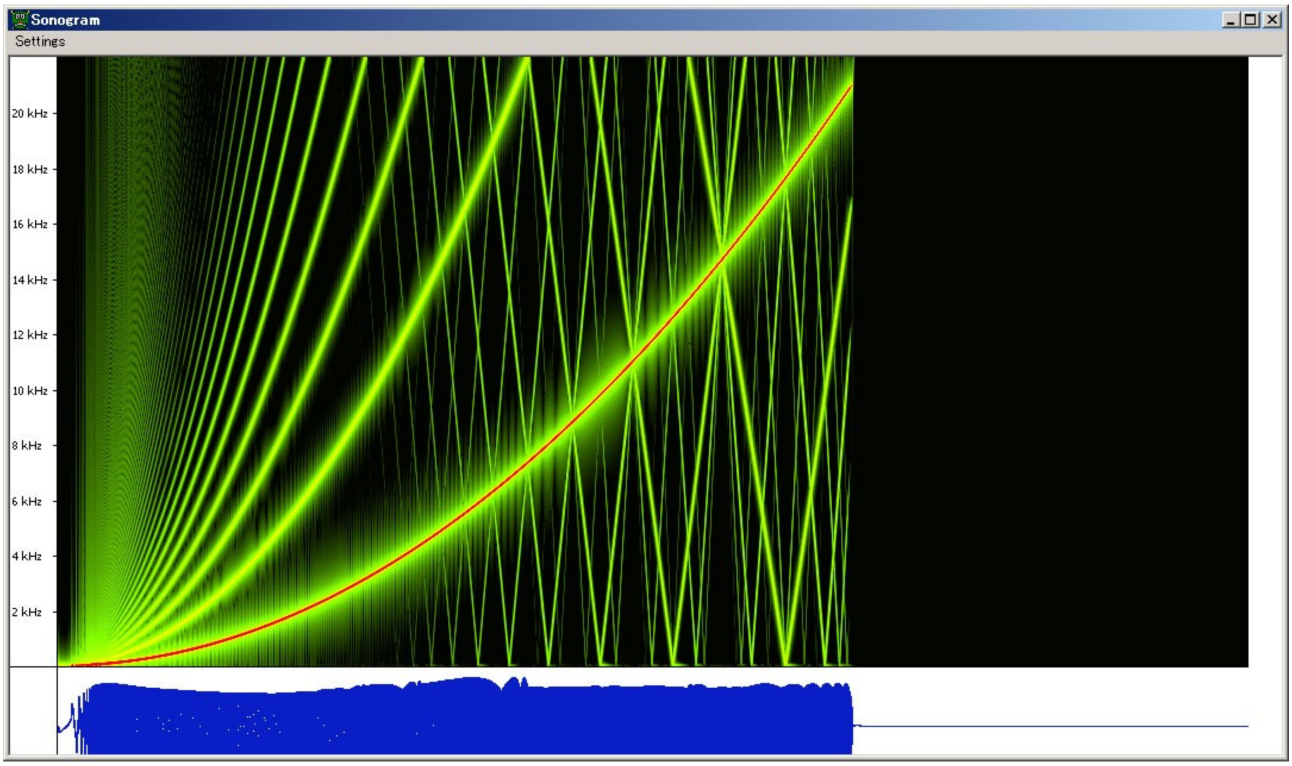



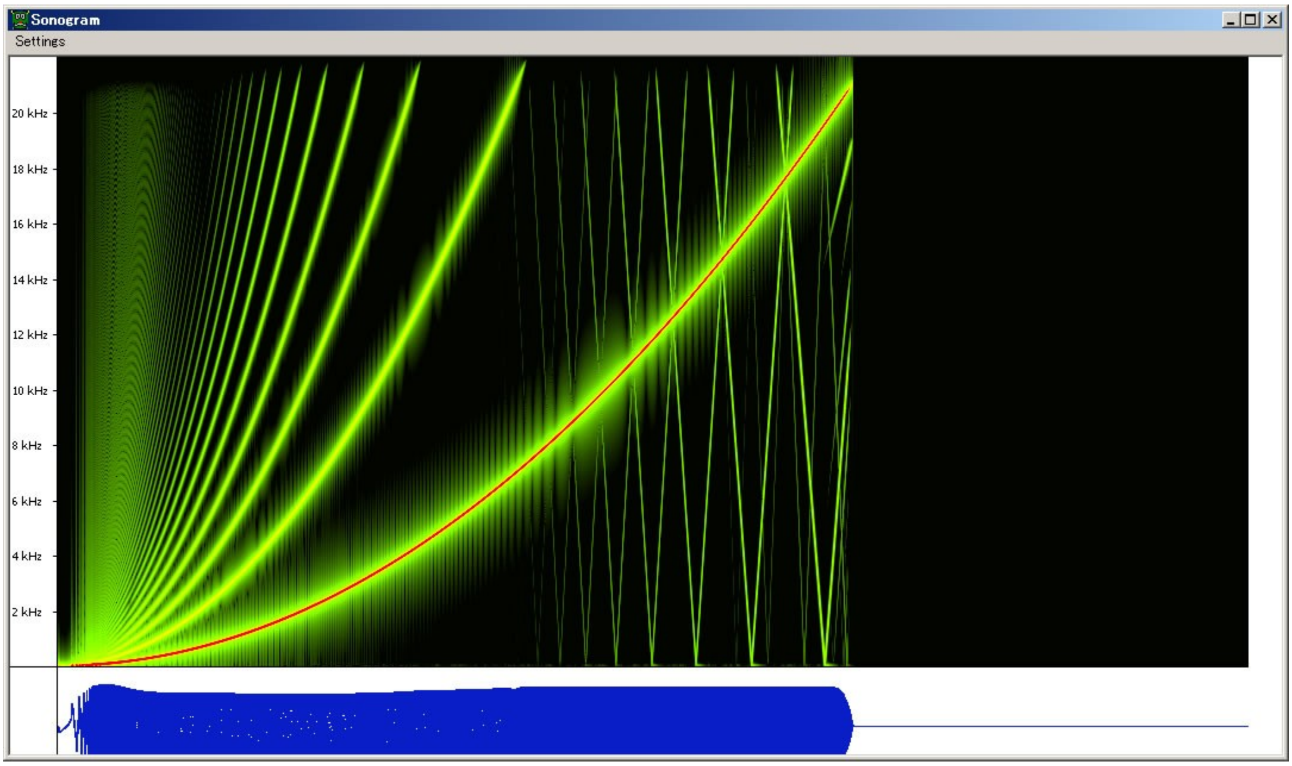



**Bootsy Tessla Pro**(Preset “Drive Max and Bass On”)

From above,  altered in 44.1kHz, 88.2kHz projects.

Description:
Fold overs are seen in both S.R. settings, but more present at lower S.R. settings.

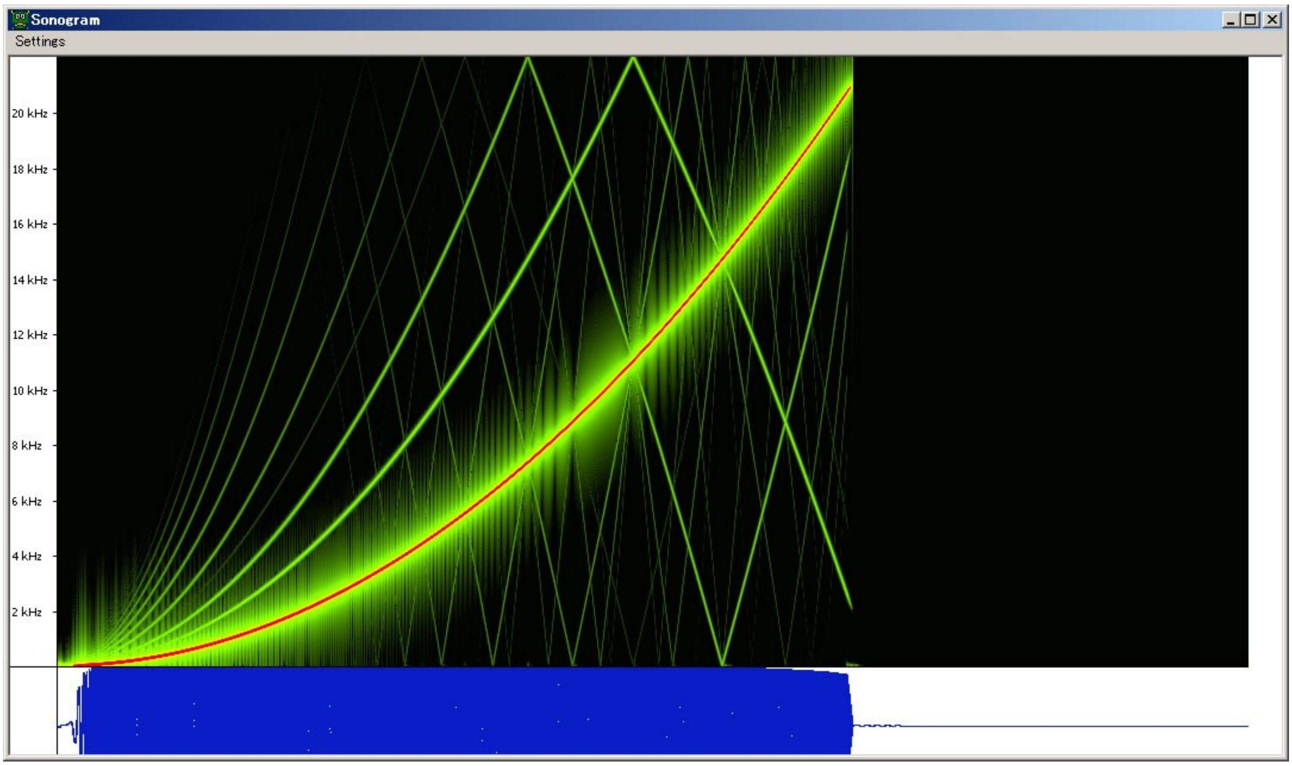


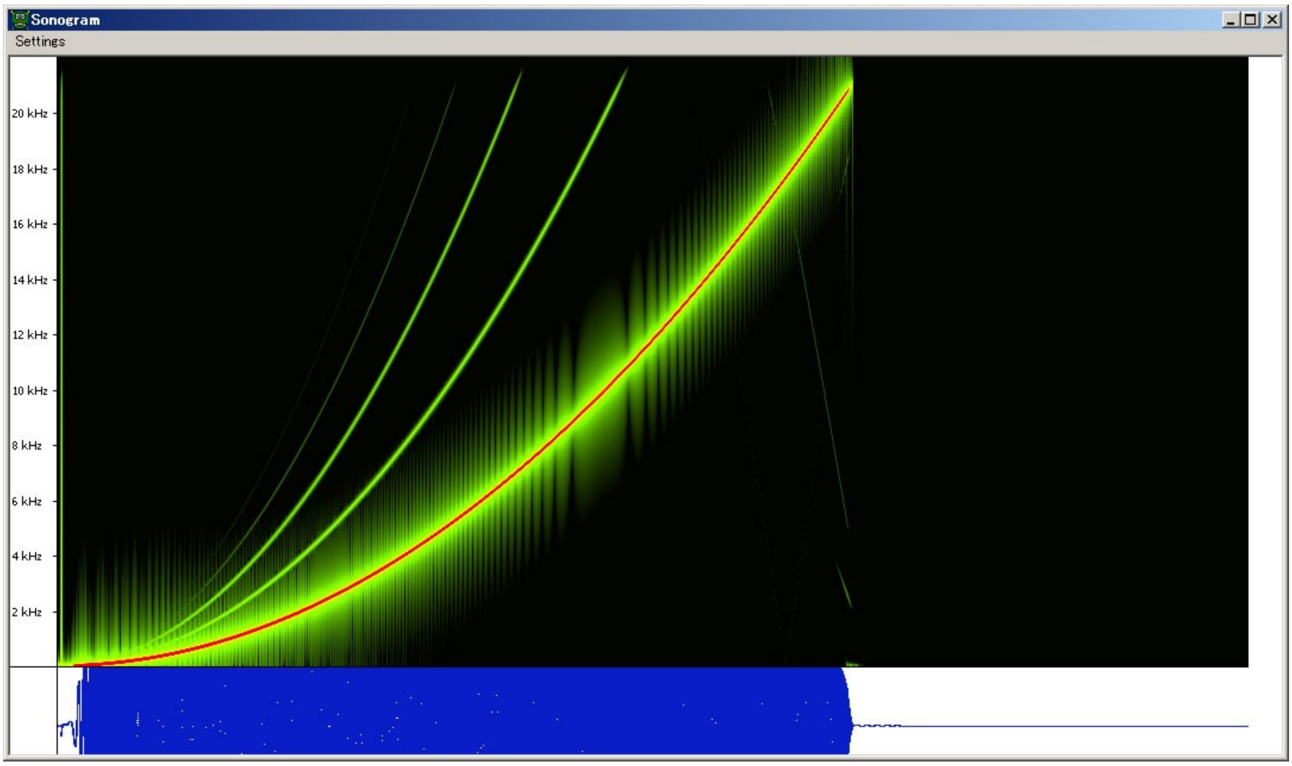



**IK Multimedia T-Racks3 / Model670** (Preset "Warm670")

From above, altered in 44.1kHz, 88.2kHz projects.

Description:
This is to examine a product by yet another manufacture.
However, T-Racks3 Deluxe suite also offers an option to internally up-convert the signal before applying the effect. Although not tested, this feature (and other products with "over sampling" enabled) is expected to reduce these foldovers.

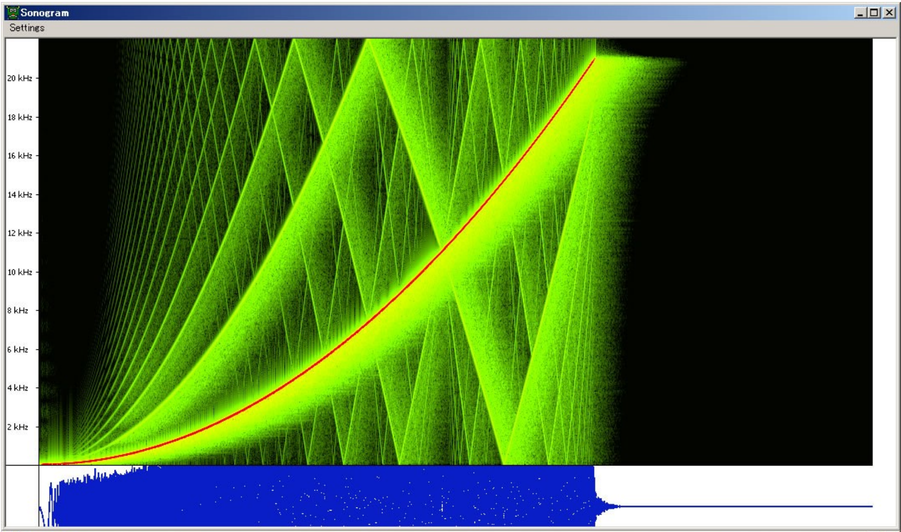


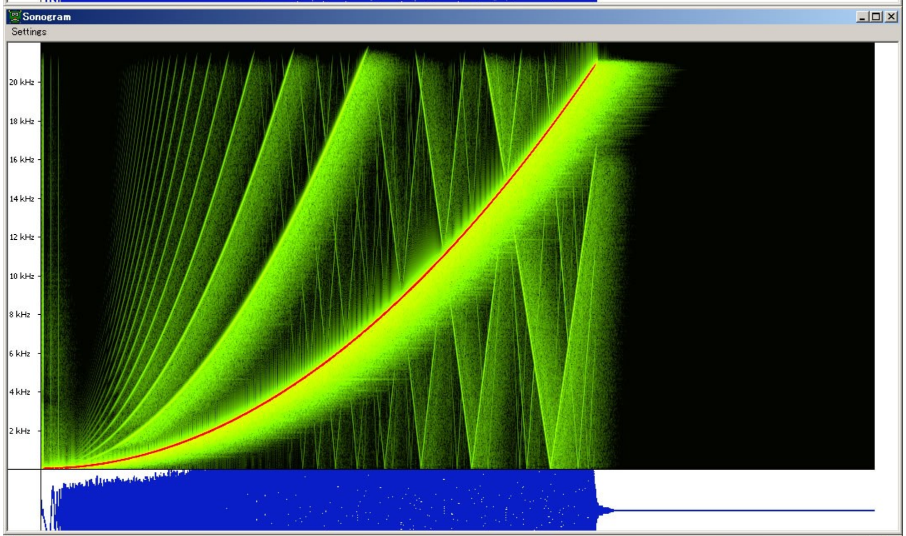



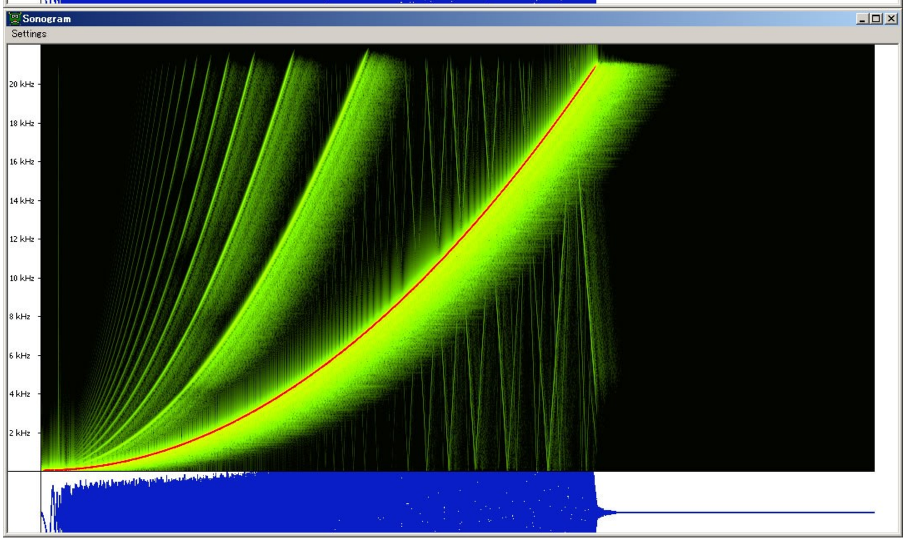
**Cakewalk VX64** ("Sugary Sweet") → **IK Multimedia CSR** (Preset "Thick Vocal Plate")

From above, altered in 44.1kHz, 88.2kHz, 192kHz projects.

Description:
Here is an example of a practical effect chain often used on vocals. The signal is sent to a saturator, then to a reverberator. A reverberator, in short is a large cluster of delays. This series of delay lines add tails to the original signal.. the forged signals are no exception, and is a possible source of (unpleasant) additional smearing.



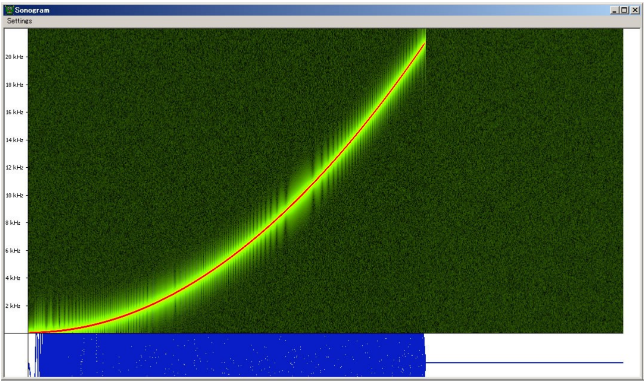



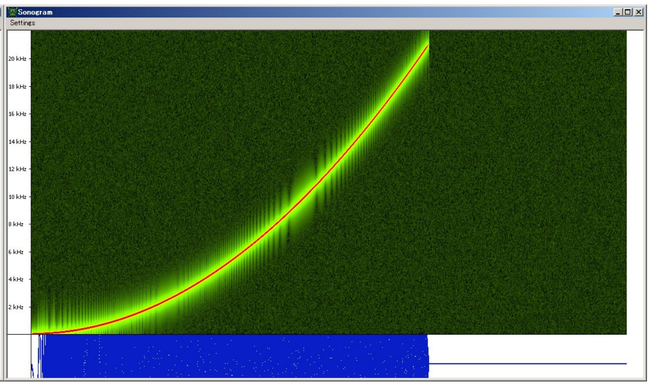
**Extra**

Comparing dither methods offered in Sonar.

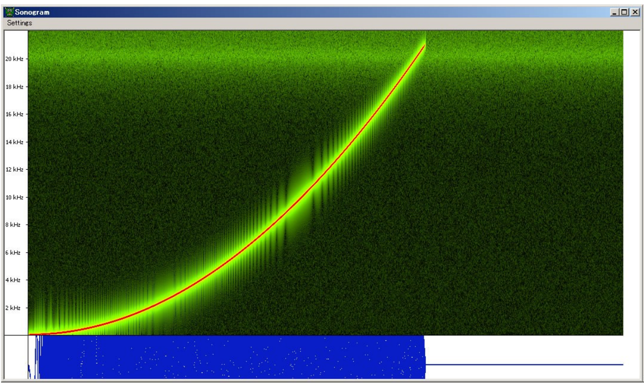
"Rectangular" and "Triangular" add noise distributed equally at all frequencies and have no visible differences. Unlike these two, "Pow-r1", "Pow-r2", "Pow-r3" each have their characteristic in the range the noise is weighted. Also, "Pow-r2" and "Pow-r3" appear to add less noise while the input is silent. The manual recommends Pow methods to be used at the final stage of mixdown, perhaps because of their relative clearness in frequency ranges the root of most instruments reside. On the contrary, the manuals suggest not to use them on material which may be edited later, perhaps because when applied multiple times, the weighted noise accumulates more easily.



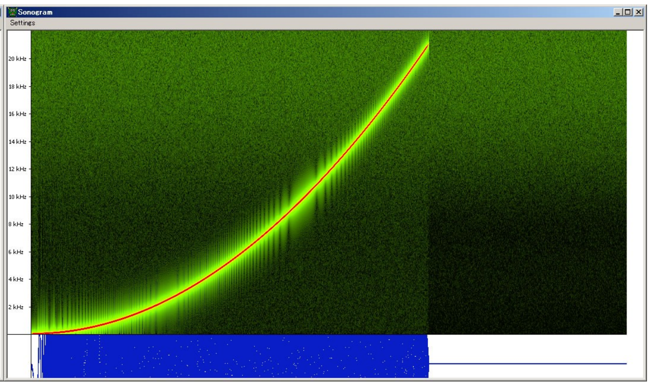
Triangular



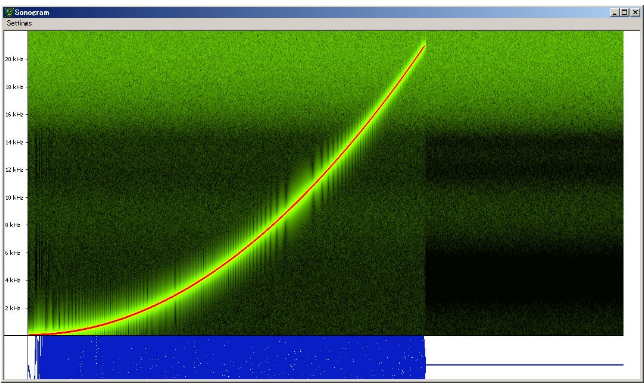
Rectangular



Pow-r1



Pow-r2



Pow-r3

Studio Gyokimae
http://pspunch.com/pd/
Rev.1 (Apr.22, 2011)