THE EVOLUTION OF THE PRE-AMPLIFIER

How the Linn DSM improves on traditional analog designs

LINN

Synopsis

In this white paper, we'll show how Linn DSM players perform the core functions of a pre-amplifier – volume control and source switching – in a way that produces a more accurate result than a traditional analog design.

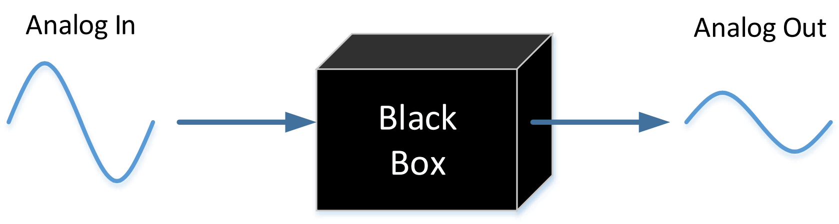
First, we'll provide an overview of noise and distortion in both analog and digital systems.

Next, we'll compare measurements of Linn's Klimax DSM to the Klimax Kontrol and show how the technology used in the Klimax DSM substantially reduces both noise and distortion.

Finally, we'll discuss the complete signal path of the Klimax DSM and explain its benefits.

1. The pre-amp as a 'black-box'

An analog pre-amp has two simple jobs to do: source selection and volume control. How this is achieved is largely irrelevant. What *is* important is what it does to the audio signal. To this end, let's consider the pre-amp as a 'black-box' system:

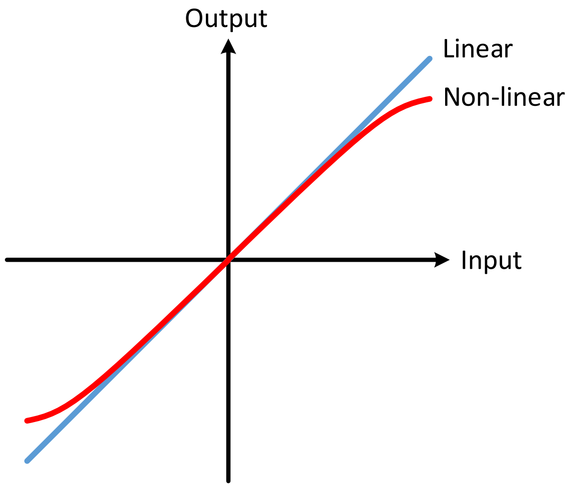


We can evaluate the performance of any black-box system by passing a signal through it and observing the output. Does it distort the signal? Does it add noise to the signal?

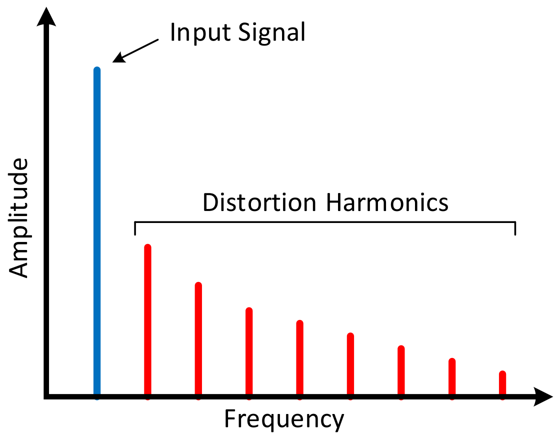
A perfect pre-amp would do nothing other than change the signal level. Unfortunately, in the real world, passing an analog signal through any electronic circuitry (active or passive) will add noise and distortion. As designers, all we can do is try to minimise the damage.

Distortion

Distortion is generated when the transfer function of an electronic component (output vs. input) is non-linear.



A non-linear transfer function distorts the original signal by generating new signals at multiples of the input frequency. These new signals are called harmonics, hence the term 'harmonic distortion'. Harmonic distortion is easily identified by the presence of unwanted frequencies in the output spectrum.



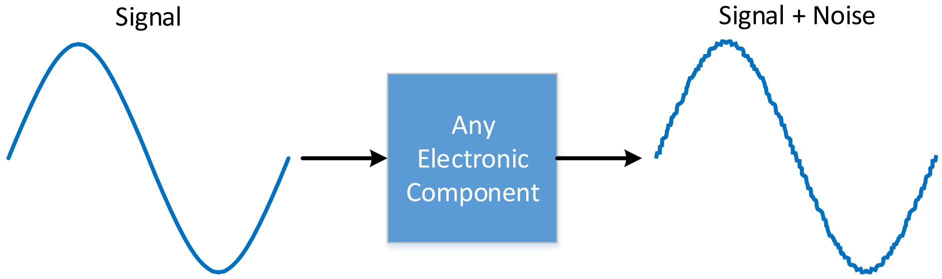
All active analog components (e.g. diodes, transistors, op-amps, etc.) and some passive ones (e.g. transformers, capacitors, inductors) will generate distortion to some degree. In an analog pre-amp, the primary sources of distortion are usually the switching transistors used for source selection and attenuation (volume stepping).

Distortion can be reduced, and in some cases eliminated, by replacing analog processes with digital ones. Digital source switching and attenuation, for example, can be made perfectly linear, although the conversion from analog to digital (and back again) will always add some distortion due to the analog components used in the conversion circuits.

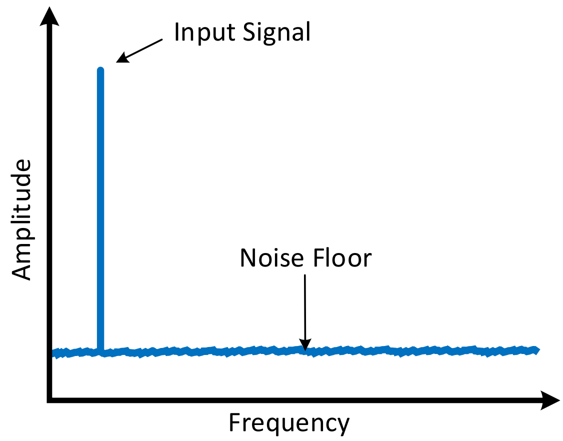
Noise

Noise is a random signal generated by fundamental physical processes within all electronic components. Even a resistor, perhaps the simplest of electronic components, generates noise just by existing at room temperature.

In an analog system, each component in the signal path will add some noise to the signal. Noise performance can be improved through careful circuit design and component choice, but no analog circuit can be completely noise-free.

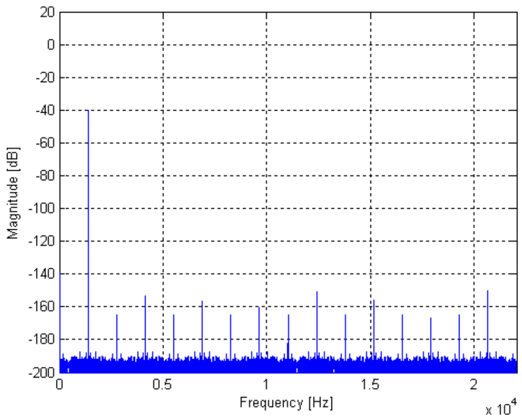


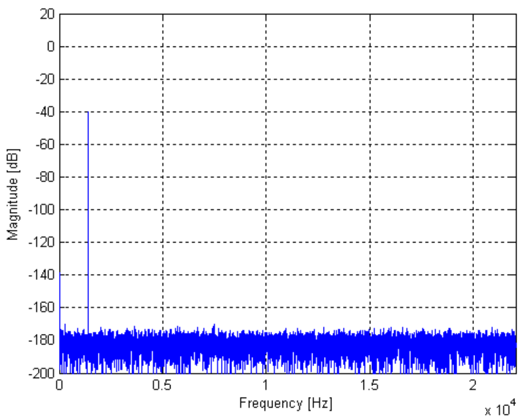
The level of noise can be measured either by looking directly at the signal in the time-domain (as above), or alternatively, by observing the noise 'floor' in the frequency domain.



Noise exists in digital systems too, but the noise sources and coupling mechanisms are different to those found in analog systems. The transistors used in digital circuits all generate noise (they are fundamentally analog components after all), but the level of noise they generate is well below that required to cause the digital signal to change state (e.g. flip from a '1' to a '0'). A digital signal does not therefore accumulate noise just by passing through a digital circuit.

A particular form of digital noise is generated whenever a digital signal is re-quantized. This happens in most DSP processes whenever the word width (i.e. the number of bits used to represent each sample) is truncated. The artefacts generated by this truncation process are called 'quantization noise', or perhaps more accurately 'quantization distortion' since it is not a truly random process. Quantization distortion can be dealt with effectively by a process called 'dithering'. This randomises the quantization process and replaces the quantization distortion components with a benign, low-level noise floor.





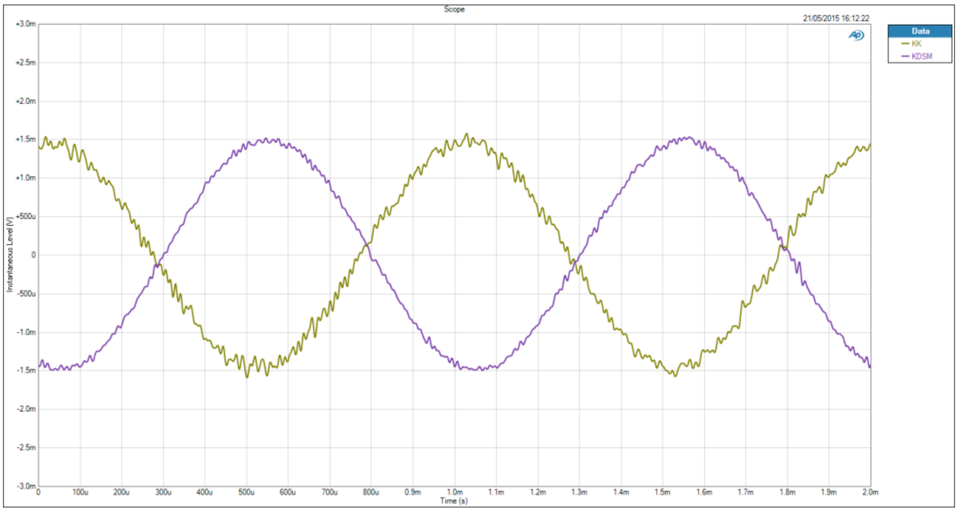
The graphs above illustrate the effectiveness of the dithering process. The left-hand graph shows the spectrum of a sinusoidal signal that has been attenuated and re-quantized to 24-bits. Quantization artefacts are clearly visible as a series of low level harmonics. The right-hand graph shows the effect of adding dither during the quantization process. The distortion artefacts have disappeared, and we are left with a low-level noise floor (about 1/10th of the noise generated by a good op-amp circuit).

2. Two pre-amps compared

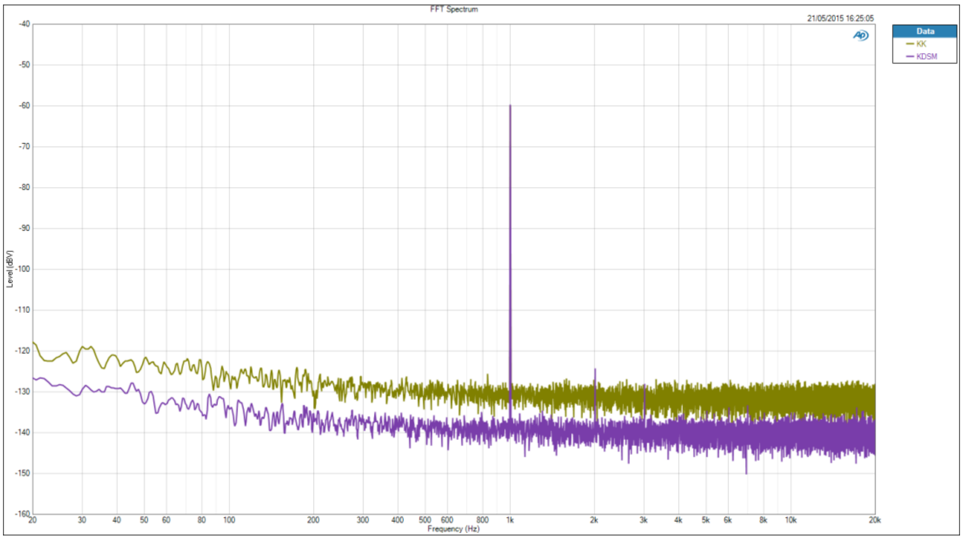
Using our black-box approach, we can compare two pre-amps by feeding them identical analog signals and observing their respective outputs.

The two pre-amps under test here are the Linn Klimax Kontrol and the Linn Klimax DSM. Input and output is via XLR in both cases, and the DSM is set to digitise the analog signal and use digital volume control.

For the first test we feed in a 1kHz sinusoidal test tone at $1V_{rms}$ and adjust the volume on both units to attenuate this to a rather quiet $1mV_{rms}$ (-60dB). The output signals are shown below - the Kontrol in green, the DSM in purple.

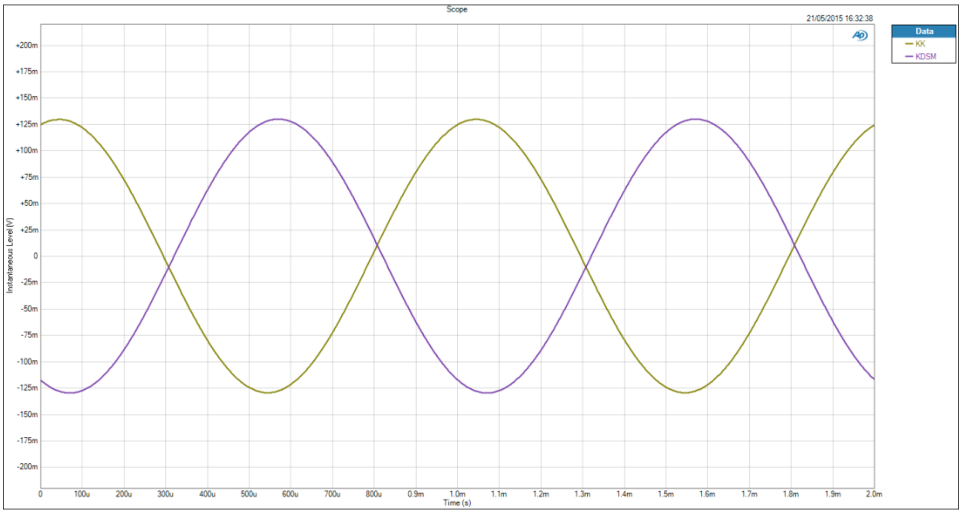


Even at this low volume setting the purple DSM output appears to be a faithful analog representation of the input signal, with no sign of any ‘digital’ artefacts. The DSM also appears to add less noise to the signal than the Kontrol. This is confirmed by looking at the same signals in the frequency domain.

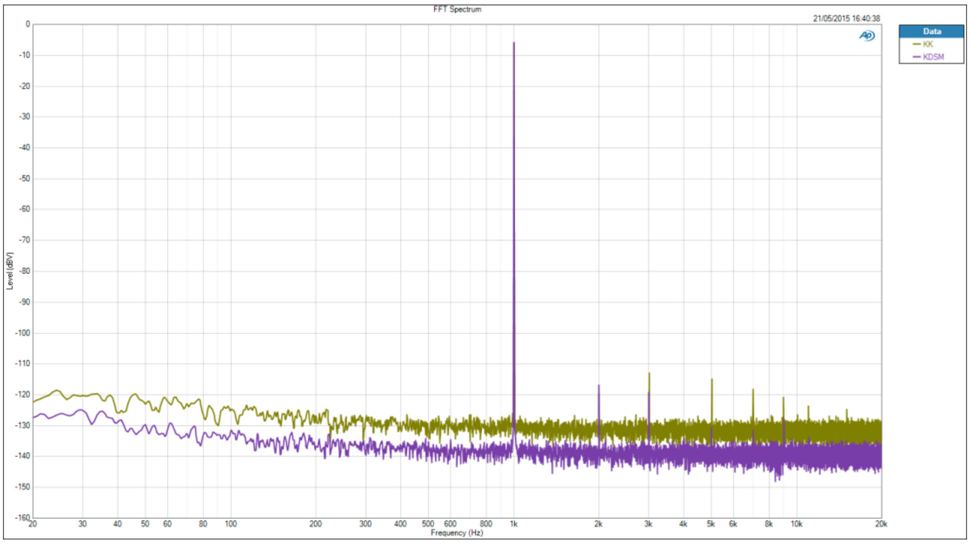


The purple noise floor of the DSM is some 8dB lower than the Kontrol across the audio band.

Next we increase the volume until the output signal reaches a rather loud 500mV$_{rms}$. This larger signal should begin to generate some distortion.



No distortion is visible in the time domain signal, but we can clearly see some harmonic products in the frequency domain.



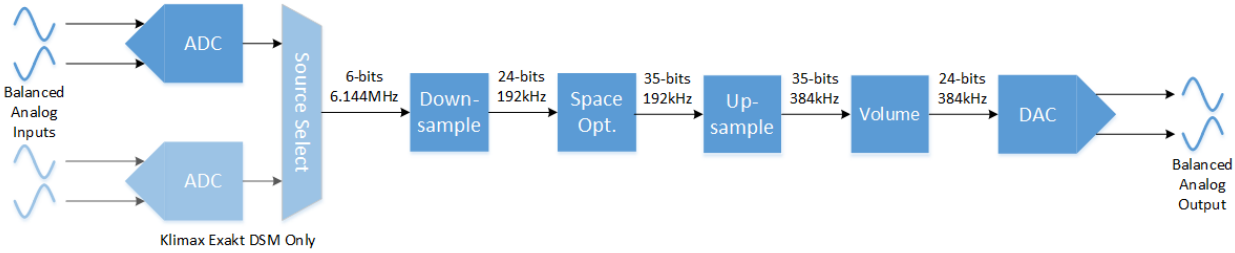
Distortion levels at the 2nd harmonic frequency (2kHz) are roughly equal between the two products, but the Kontrol generates higher levels of all the higher order products. High order distortion is generally considered to be more objectionable than low order distortion.

We can conclude from these measurements that the Klimax DSM does less damage to the analog signal than the Klimax Kontrol - less noise is added, and the output is less distorted.

But how is that possible given all the digital stuff that goes on inside the DSM?

3. A journey through the Klimax DSM signal path

The Klimax DSM signal path for analog inputs is shown below:



Analog to digital conversion

The analog signal passes through minimal analog circuitry before being digitised by a high performance analog to digital converter (ADC). The performance of the ADC in the Klimax DSM actually *exceeds* the performance of the DAC, ensuring that the analog signal is captured transparently.

Digital source switching

The Klimax DSM has only one analog input, so source switching is not an issue, but the Klimax Exakt DSM has two. Rather than switch in the analog domain where distortion and noise would be added, the Klimax Exakt DSM provides separate ADCs for each analog input. This minimises the analog path for each input and moves the source switching operation to the digital domain where it can be performed without loss.

Down-sampling

The ADC is an oversampled sigma-delta converter, meaning that its initial conversion runs at a very high rate (6.144MHz). Normally, the ADC would use its own internal logic to convert this to a more manageable sample rate, but in the DSM we are able to bypass this process and implement it instead in a custom designed FPGA. The benefits of doing this are twofold: analog performance is improved since there are now fewer noisy processes going on inside the ADC chip, and digital performance is improved due to the vast processing power available in the FPGA.

Space Optimisation

One of the benefits of a digital pre-amp is that it enables the use of corrective algorithms such as Space Optimisation that would simply not be possible in the analog domain. Space Optimisation processing is not mandatory, it can be turned off, but it can bring huge improvements to the way a system integrates with its physical environment.

Up-sampling

As with the ADC, this noisy and lossy process in the DAC is bypassed and implemented instead using a high precision algorithm implemented in an FPGA. Again, this allows for improved digital and analogue performance in a similar way to the ADC.

Volume control

The volume control module is placed at the end of the digital signal path, just before the DAC. This ensures that all intermediate processes are performed with optimal precision.

The volume adjustment process is performed by simple multiplication. If we want to reduce the signal level by 3dB we multiply each sample by 0.707. If we want to reduce it by 6dB we multiply by 0.5, and so on. This can all be done losslessly as long as we keep all the bits resulting from the multiplication process. This creates a problem, however, since the result of this multiplication exceeds the word width required by the DAC. Simply truncating each sample to match the DAC word width would generate unwanted quantization artefacts. Instead, the DSM volume control uses the dithering process described earlier to transform the truncation artefacts into a benign noise floor.

The noise floor generated by a digital volume control is no different to the noise floor generated by an analog volume control. In fact, as we have seen, the noise floor of a carefully designed digital volume control can be lower than that of the best analog designs.

In both cases, the noise floor remains constant when the volume level is reduced. This is an unavoidable consequence of reducing the signal level. So, while it is true that the signal-to-noise ratio of a digital volume control reduces at low volume settings, the ***same*** is true for an analog volume control.

The main advantage of the digital volume control is that it is completely free from distortion. The same cannot be said of an analog volume control as we have seen from our measurements.

Digital to analog conversion

The digital-to-analog conversion stage is the main source of noise and distortion in the signal chain, so great care must be taken in its design. The Klimax DSM uses the same transformer-coupled DAC stage as the Klimax DS, with its ultra-low noise reference voltage, ultra-low jitter clock, and distortion-cancelling transformer drive circuit. The transformer-coupled output gives the added benefit of galvanic isolation from downstream equipment.

An end to digital vs. analog?

A pre-amp is a pre-amp, be it digital or analog. Digital processing and analog circuits are just the technological means of achieving the functionality one is after; in the case of the preamp, source switching and volume control. What matters is not the technology chosen, but how well that technology allows one to protect the music signal.

Of course, measurements and engineering analysis can help us understand ***why*** something sounds better, but listening is the only way to confirm that it ***is*** better.

Through extensive Tunedem based listening we are confident that the Klimax DSM is the best pre-amp Linn has ever made.

More information about the Klimax DSM is available at www.linn.co.uk/klimax-dsm