Analog Design
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Oversampling

[Back to Contents](#)

Overview

This paper rectifies a very common misunderstanding as to the nature of oversampling. That is, the principle of sampling a signal higher than the Nyquist rate, and suitably processing the results, results in a larger signal to noise ratio. The key bit being that the *final S/N ratio* is improved. The noise itself, as often misstated, does not become lower in density. That concept is simply a mathematical abstraction, only meaningful when *additional conditions* are applied in the mathematical universe, with no basis in physical reality.

The physical basis of oversampling is that of adding up a signal N times, and adding up the noise N times. If the noise is random, "adding" up the noise will only result in an increase as sqrt(N), hence the S/N increases as sqrt(N),

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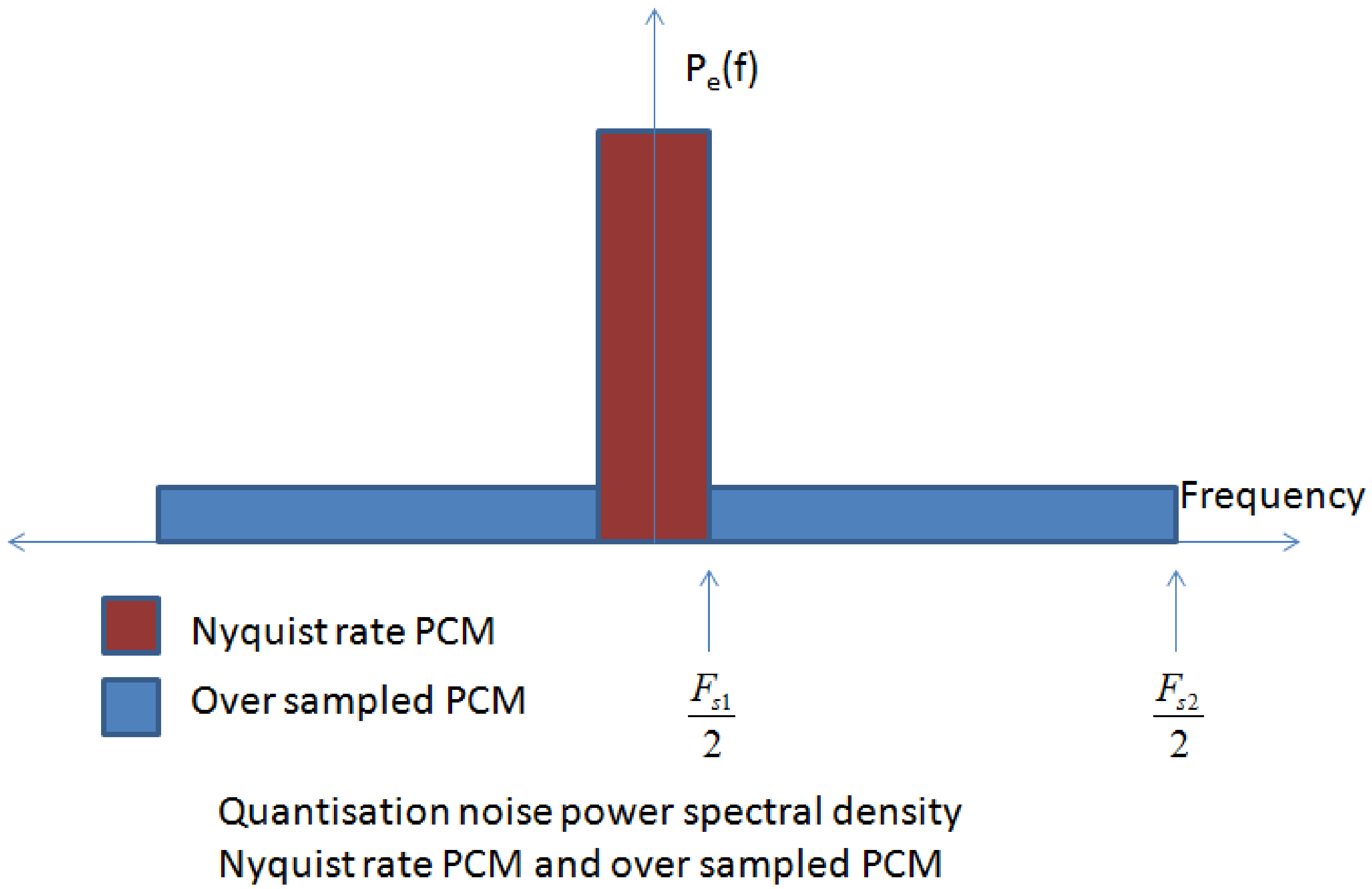
Introduction

The technical literature abounds with the notion that sampling at a higher rate, magically results in the actual sampled noise density at a sampling instant being reduced. It is commonly expressed for the following diagram:

Fig. 1



A typical accompanying explanation of this diagram is usually rather vague and states stuff along the lines of "as the original noise is spread about $F_{s1}/2$ this same power must now be spread about $F_{s2}/2$, and hence its density must be reduced". Usually, little attempt is made to explain any more details.

So, apparently, merely measuring a dc voltage twice, rather than once in the same time, results in each individual measurement being reduced in value? Ah…hands up those that see something rather fishy going on here?

It is of course, all complete nonsense, and easily disproved by running spice simulations rather than pontificating about with high brow mathematics. Noise is no different in the sense of it being a voltage as any other signal. If noise gets reduced by sampling more often, so must the signal!

Voltage signals are not mathematically abstractions. They are physically real. They can be measured. Although it is mathematically valid to manipulate symbols and reformulate equations with redefined variables, these new variables should not be confused with physical reality. If a real measurement is made on a system, the measurement, if correctly designed, will not depend on how often the measurement is made. If is does, the measurement is in error.

Oversampling Schematic

Fig 2.

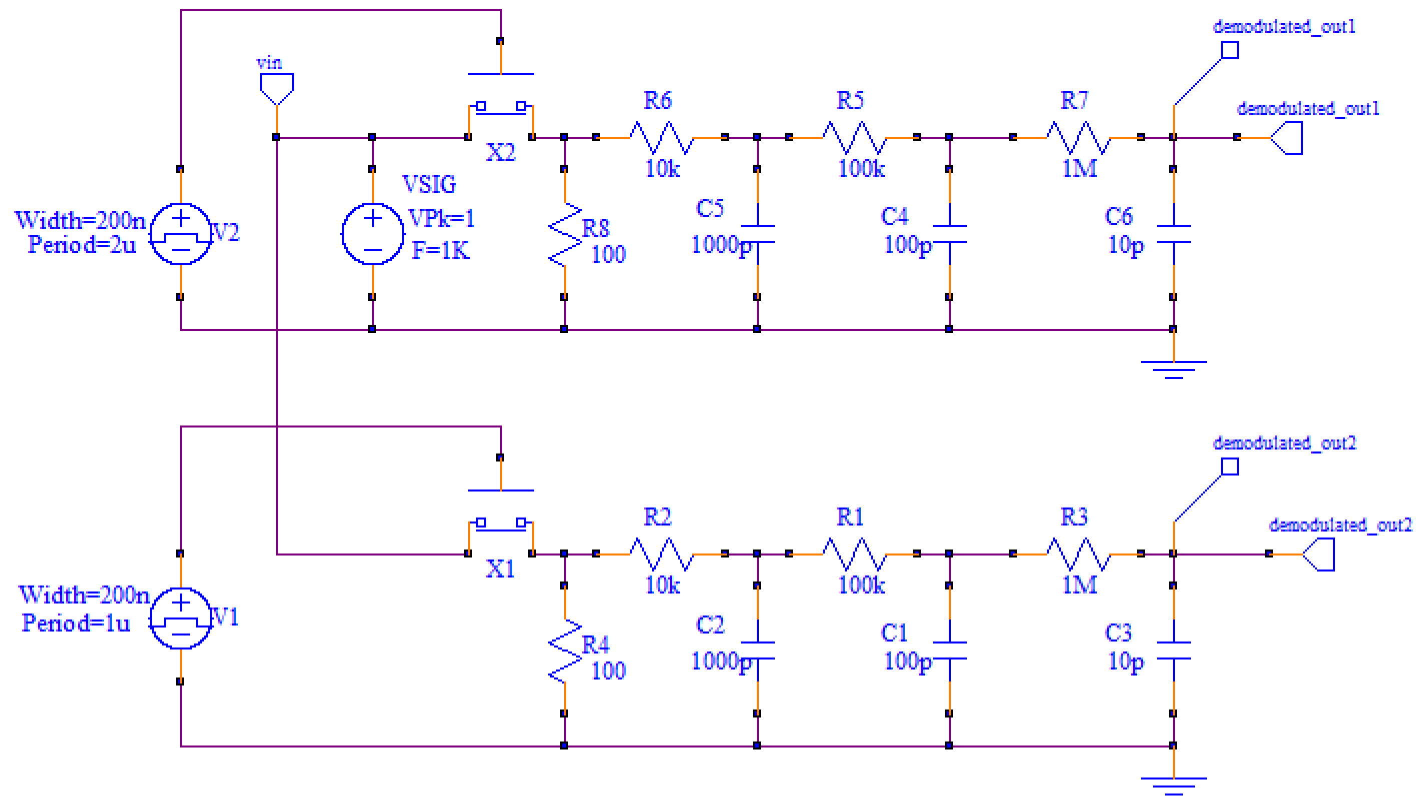


Fig. 2 represents a real physical embodiment of the sampling process, by way of PAM. V_{in} is a sine wave signal, applied to a sampling switch (X), and then filtered with a low pass filter. The signal is sampled at two different rates. The Spice results for the schematic are:

Fig. 3 Demodulated Waveforms

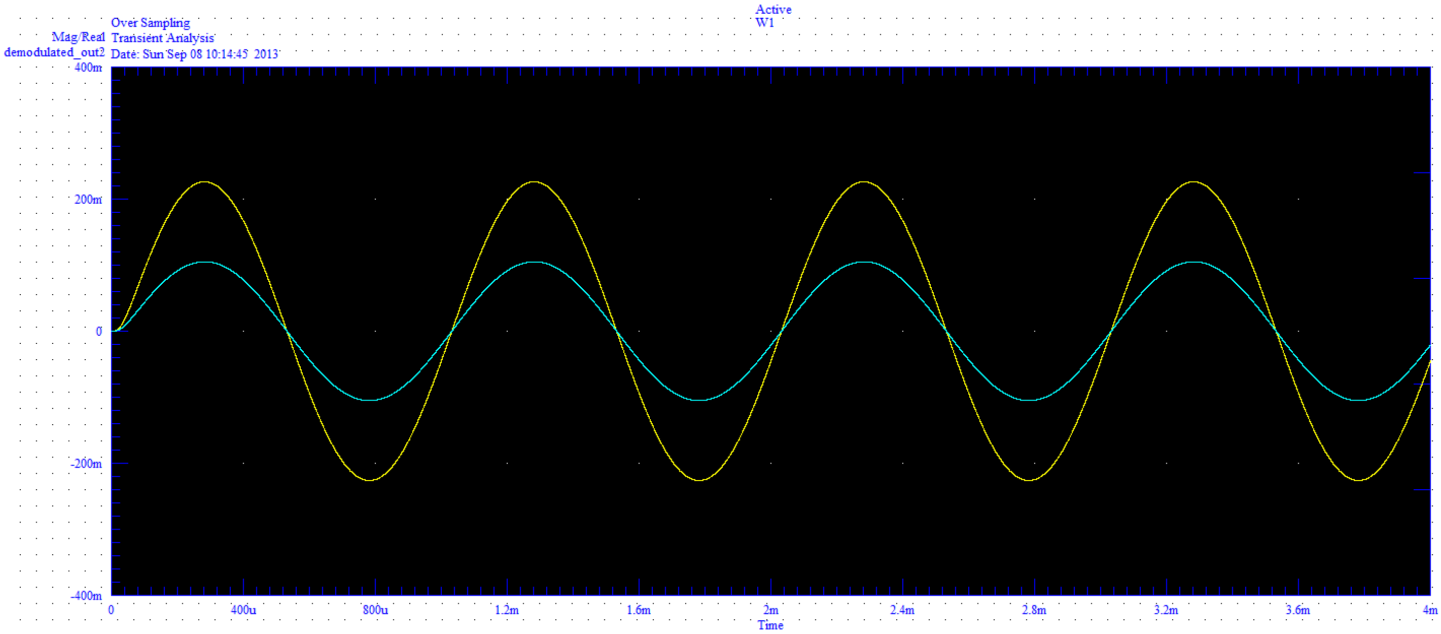
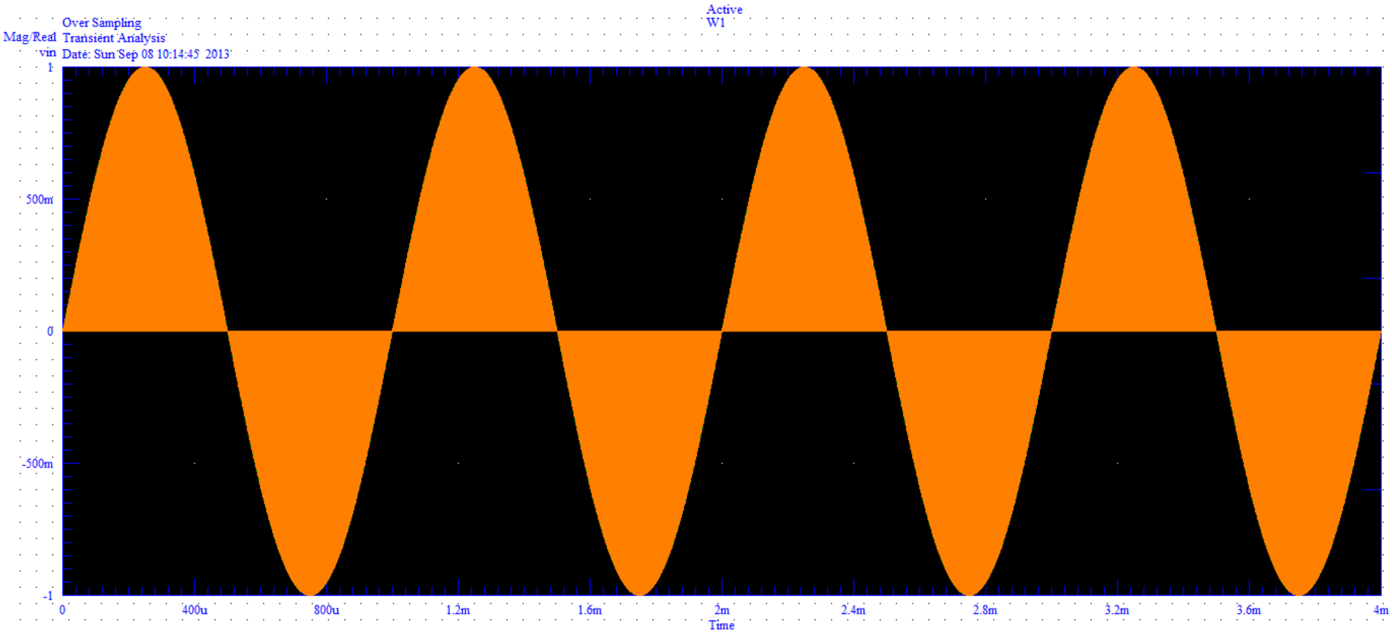


Fig. 4 Modulated Waveforms



Analysis

The demodulated output waveforms show that the signal being sampled with twice the sample rate of the other, has twice the output. For a noise signal on top of the sine signal, sometimes the noise would add, and sometimes the noise would reduce the signal amplitude at the same equivalent sampling time point. By the usual statistical mathematical arguments, it is anticipated that the reader should have no problem in accepting that the net effect is that the sum of the noise, without the signal, increases as sqrt(N), not as N, and that is left for them to confirm at their leisure.

In a real ADC system signals are digitally filtered, i.e. low pass filtered which *adds up all the samples*. If it were true that the actual noise density was reduced by sampling faster, after the signals were sampled, most could then be simply thrown away before any filtering was done! This is of course, never done. It would achieve nothing. *After* the signals are filtered, it is then perfectly fine to throw away some of the samples in the process known as decimation. It is only by adding up all the samples, that oversampling can work, hence to state that noise density is *reduced* is false.

It should be noted that, in real digital PCM systems, the noise is not actually noise. *It is fully deterministic distortion*. For repetitive signals, it is only the principal that the sampling instants are not correlated or locked with the sampled signal that prevents the signal being sampled at the same point in its waveform, and appear "as if" the distortion is random noise. If the signal is locked to the sampling instant, oversampling does nothing. The error in the waveform will always be the same.

Conclusion

Oversampling does not result in a reduction in noise density, the noise density in fact remains, exactly the same. It actually results in an *increase* in the total noise when all the sampled signals are passed through a filter. However, this noise does not add up in the same manner as that of a slowly changing signal waveform. The demodulated (low pass filtered) signal waveform amplitude adds with the number of samples, and the noise adds as the sqrt(N) so that the final S/N ratio is improved by a factor of sqrt(N).

References

1 – "An Overview of Sigma-Delta Converters: How a 1-Bit ADC achieves more than 16 bit Resolution" – Pervez M. Aziz, Bell Laboratories, Henrik V. Sorensen, Arial Corporation, Jan Van der Spiegel, University of Pennsylvania

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