

Correspondence

Comments on "The Modulo Time Plot: A Useful Data Acquisition Diagnostic Tool"

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In this paper¹, the use of a reordered sample set is very useful for quick visual inspection of system performance. This artificial oversampling algorithm unrolls spectra as well [1]. However, the question is how to reorder the samples.

To answer that, one must recall the conditions for *uniform permutation* of samples [2], [3]: the sample set of a sine wave with a period $T = 1/f$ and sampling frequency $f_s = 1/\Delta t$.

$$x[i] = \cos(\omega i \Delta t) = \cos(2\pi i f / f_s), \quad i = 0, 1, \dots, N - 1$$

is *periodic* only if the value of numerical (normalized) frequency

$$f/f_s = m/N, \quad 1 \leq m < (N/2)$$

where m is prime to N , and N is the numerical period: $x[i] = x[i + N]$. This is the case of *coherent sampling*; N unique samples from m periods of the signal, $N \cdot \Delta t = m \cdot T$. Such a raw sample set ($m > 1$) can be reordered into a *single* period by an index transformation

$$j = (Ji) \bmod N$$

where J is a unique integer and must be calculated from the relation

$$(mJ) \bmod N = 1$$

i.e., J is multiplicative inverse of m .

Fig. 1 shows a simple but informative example of transposition, where $m/N = 7/32$, so that $J = 23$. Some short notes on the simulation:

- 1) use DFT if N is not a power-of-2;
- 2) n may be noninteger, and then it is the effective number of bits;
- 3) the common *sinc* interpolation formula [5] can be rearranged into a *finite sum* in case of *periodic* sample sets;
- 4) spectrum levels are in dB (voltage ratio);
- 5) with $N = 32$, values of J for other possible frequencies

$$m = 3 \quad 5 \quad 9 \quad 11 \quad 13 \quad 15$$

$$J = 11 \quad 13 \quad 25 \quad 3 \quad 5 \quad 15$$

- 6) since the numeric frequency is a ratio of small integers, there is a massive superposition of the aliased harmonics.

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¹F. H. Irons and D. M. Hummels, *IEEE Trans. Instrum. Meas.*, vol. 45, no. 3, pp. 734-738, June 1996.

DISTORTION due to DIGITIZING OF SINUSOID

Mathcad

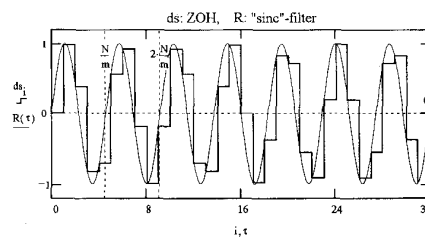
(m prime to N; N = power-of-2 for FFT)
no. of samples: N:=32 i:=0..N-1 no. of bits: n:=8 D:=2ⁿ
SIGNAL frequency: m:=7

sampling: $s_i := \sin\left(2\pi \frac{m}{N} i\right)$ [jitter-free]
quantizing: $ds_i := \frac{\text{floor}\left(\frac{D}{2} s_i + 0.5\right)}{\frac{D}{2}}$ [rounding]

reconstruction: [zero-order-hold(ZOH) and "sinc"(lowpass filter) interpolation]

$$r := 10^{-6}, 0.1..N$$

$$R(\tau) := \frac{\sin(\pi\tau)}{N} \sum_{i=0}^{N-1} (-1)^i ds_i \cot\left(\frac{\tau-i}{N}\right) \quad [N \text{ even}]$$



normalized signal period: [no. of samples per period]

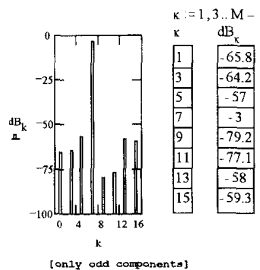
$$\frac{N}{m} = 4.57$$

SPECTRUM

Harmonic (= h.m) and ALIASED (= a) components:

c := FFT(ds) M := last(c) k := 0..M
 $dB_k := 20 \log(\sqrt{2} |c_k| + 10^{-6})$

h = 1, 3..19
 $p_h := \text{mod}(h.m, N)$
 $a_h := \text{if}\left(p_h < \frac{N}{2}, p_h, N - p_h\right)$



h	h.m	a _h	signal
1	7	7	11
3	21	21	11
5	35	35	3
7	49	49	15
9	63	63	1
11	77	77	13
13	91	91	5
15	105	105	9
17	119	119	9
19	133	133	5

Fig. 1. Mathcad simulation: spurious components due to quantization.

Simulation parameters may be easily modified, i.e., in [1] $m/N = 1501/4096$, so that $J = 2677$, or, with reindexing from zero in [5, Fig. 2] the numeric frequency is $m/N = 4/19$, so that $J = 5$.

The technique of the uniform permutation can be used for other desirable waveform or spectrum modifications.

REFERENCES

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