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Nyquist and the PMU

NASPI

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Why talk about Nyquist?

- Delay in getting data from PMU too large for HVdc control
  - "Significant fraction of a second"?
  - The *measurement* problem solved in 1968, so what is going on?
- IEEE Standard C37.118.1: filter that depends on the reporting rate
- Some connection ?
- The connection: Nyquist?

Confession

- This has been a challenging problem to study
- Much help has been provided by working group
- and by non-working-group colleagues

Even at this point,

I am not *certain* that I understand the problem

Now, on with the journey

Nyquist 101

- Harry Nyquist (1889-1976) wrote on the topic of sampling a continuous-time signal in such a way that it could be reproduced.
- His paper [1] contains the words

  > It is concluded that full knowledge of *N/2* sinusoidal components is necessary to determine the wave completely. It will be shown below that this number is also sufficient

- Name Nyquist now associated with a treatment of idea by Claude Shannon twenty years later.

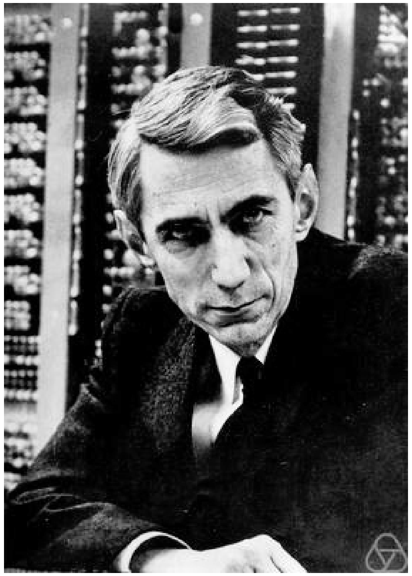
[1] H. Nyquist, "*Certain Topics in Telegraph Transmission Theory*" presented at the Winter Convention of the A. I. E. E., New York, NY, February 13–17, 1928. Printed in Transactions of the A. I. E. E., pp. 617–644, Feb. 1928.

What did Shannon add?

- Shannon said this in his exposition on sampling: [2]

  A similar result is true if the band does not start at zero frequency but at some higher value, and can be proved by a linear translation . . . of the zero-frequency case.

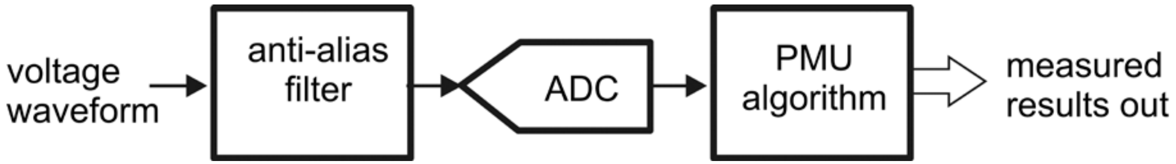
  That is, a sufficient no-loss condition for sampling signals that do not have baseband components exists that involves the ***width*** of the non-zero frequency interval as opposed to its highest frequency component.



[2] Shannon, C. E. (January 1949). *"Communication in the presence of noise".* Proc. Institute of Radio Engineers. **37** (1): 10–21. Reprinted as a classic paper in: Proc. IEEE, Vol. 86, No. 2, (Feb 1998)

Nyquist 101

For the purposes of measurement, in the case of the PMU, that usually means



Filter cutoff adjusted for the sampling rate in the A/D

But the PMU Standard mentions aliasing of the *output,* depending on the rate at which the reports are issued.

I had never heard of such a thing.

So that is the intro to the problem.

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Measuring without Nyquist

Measurement of PMU kind is a “fitting problem”

Form of equation is model of signal: $v(t) = V_{\mathrm{m}} \cos(\omega t + \varphi)$

As a fitting problem

- Need multiple samples
- Min # samples = # parameters to be fitted
- PMU equation solves with 3 samples if no noise
  - Need 4 if ROCOF

- Way below anything reminiscent of Nyquist!

PMU Measurement (2)

- AAF/Nyquist needed only because method uses Fourier transform

- Standard introduces another filter:

Out-of-band interference testing: The passband at each reporting rate is defined as $|f - f_0| < F_s/2$. An interfering signal outside the filter passband is a signal at frequency f where: $|f - f_0| \geq F_s/2$

For test the input test signal frequency f_{in} is varied between f_0 and ~~± (10%)~~ of the Nyquist frequency of the reporting rate.

That is: $f_0 - 0.1\ (F_s/2) \leq f_{in} \leq f_0 + 0.1\ (F_s/2)$

where

F_s = phasor reporting rate
f_0 = nominal system frequency
f_{in} = fundamental frequency of the input test signal

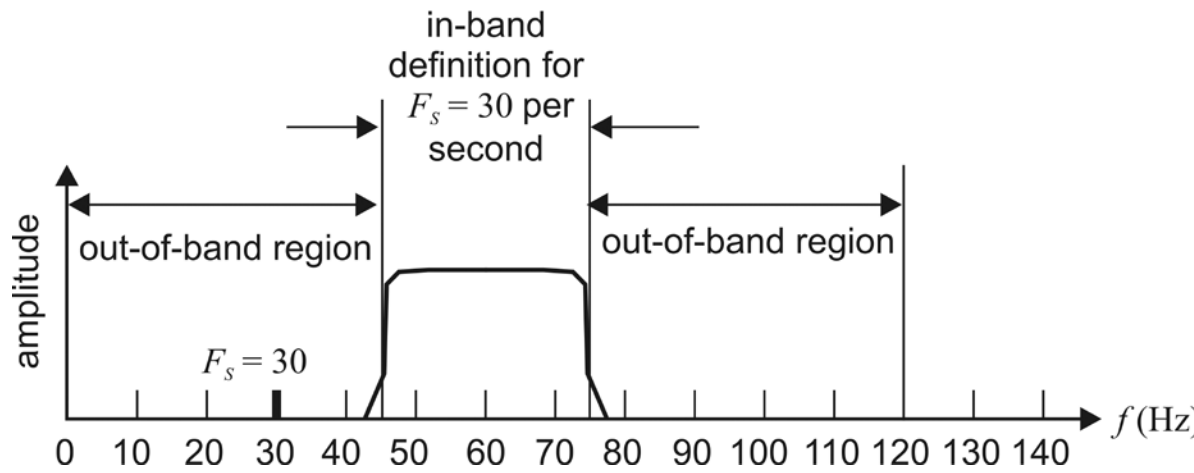
Should be "$f_0 \pm 10\%$"

- Nyquist frequency of the reporting rate?

Nyquist rate of the Reporting Rate

- I struggle with “Nyquist frequency” connected to a reporting rate

- Standard defines



NOTE 3—Compliance with out-of-band rejection can be confirmed by using a single frequency sinusoid added to the fundamental power signal at the required magnitude level. The signal frequency is varied over a range from below the passband (at least down to 10 Hz) and from above the passband up to the second harmonic ($2 \times f_0$).

- Note that this test signal avoids passband

- What does the test signal look like?

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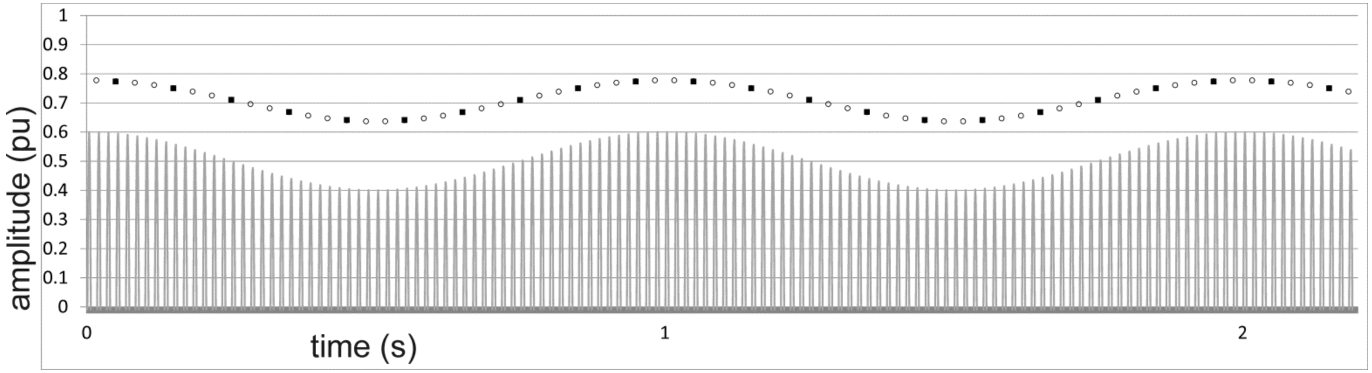
Test signal

- 61 Hz test signal – 1 Hz “beat” evident

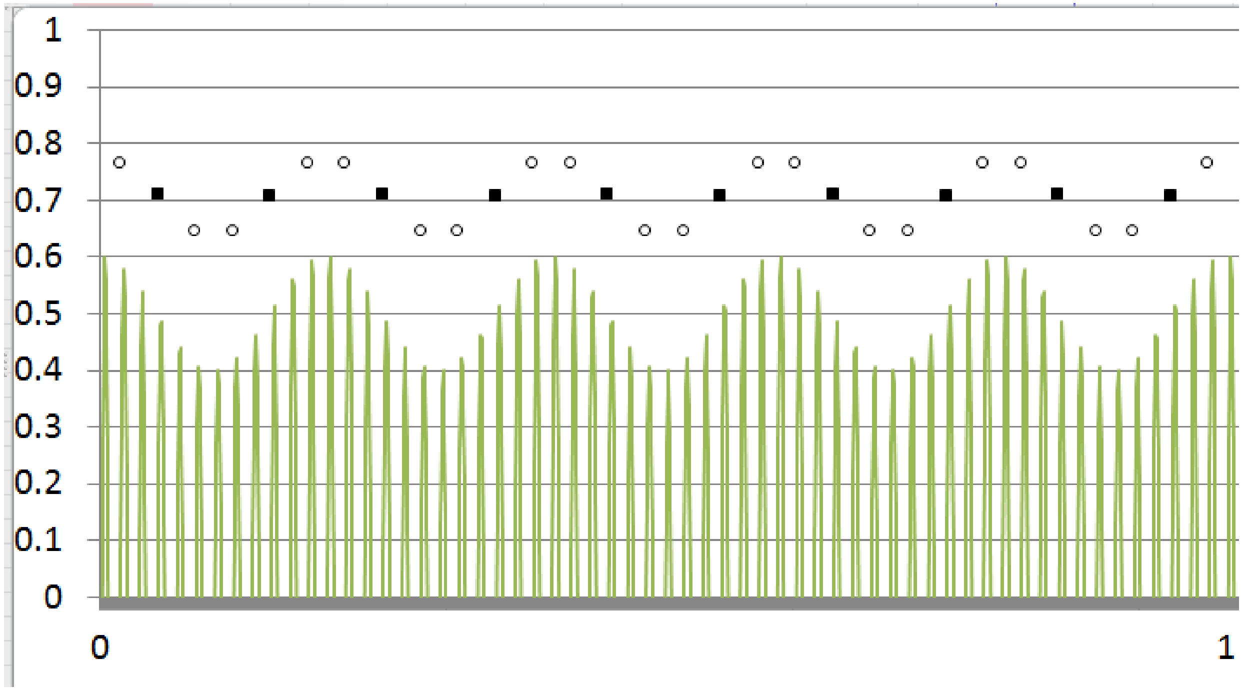


Open circles: 30 reports/s

Filled squares: 10 reports/s

Test signal

- 65 Hz test signal – 5 Hz “beat” evident

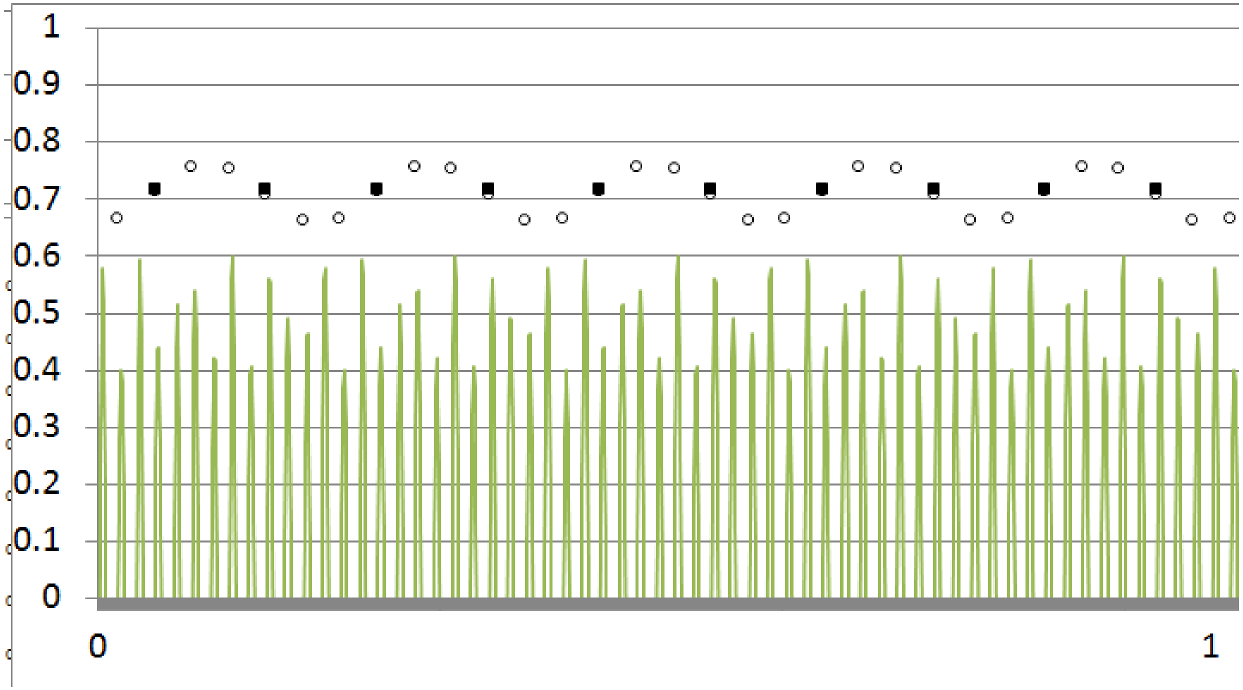


Open circles: 30 reports/s

Filled squares: 10 reports/s

Test signal

- 85 Hz test signal – “beat” evident – outside passband



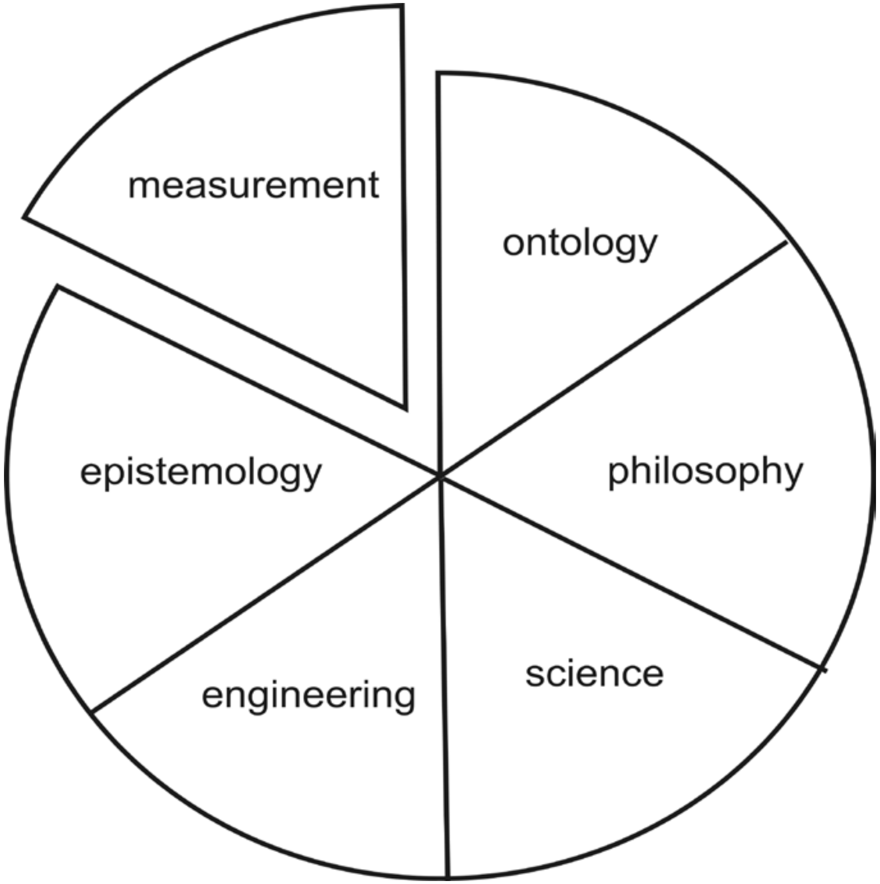
Open circles: 30 reports/s

Filled squares: 10 reports/s

Is this the issue?

- Hard to (mentally) “unscramble” reports with high beat frequency

- After much discussion and much exploring . . .



- Still not sure this is the problem

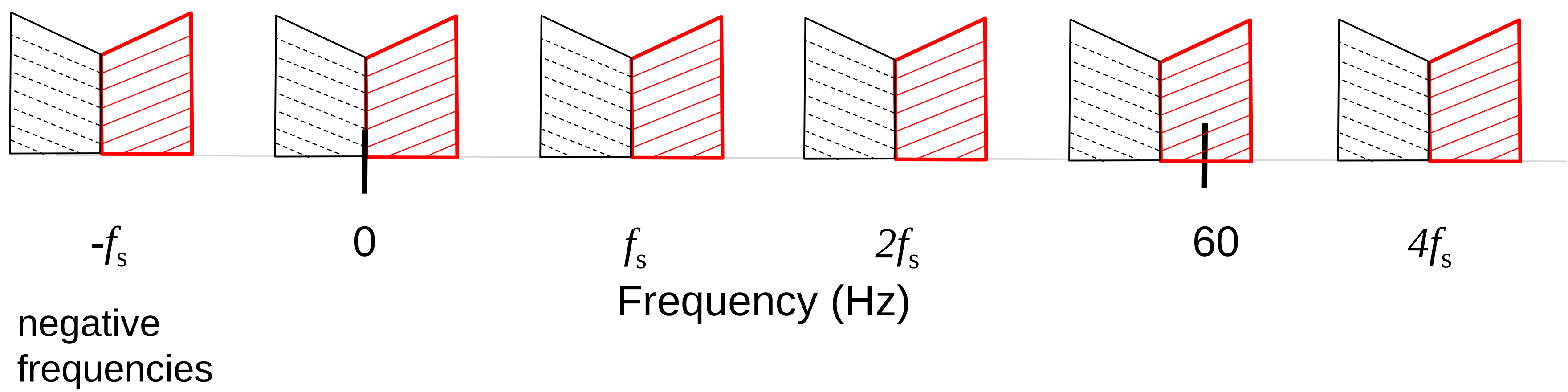
Is *this* the issue?

- It turns out Shannon did not get it all correct:
- or at least he did not get it all spelled out
- If you add a bandpass filter (as the Standard does)
  - It matters where the passband is
  - and what the sampling rate is

[3] Vaughan, R.G, Scott, N.L., White, D.R. "*The Theory of Bandpass sampling*" IEEE Trans Signal Processing, Sept 1991, **39** (9), pp. 1973--1984

Here's the problem

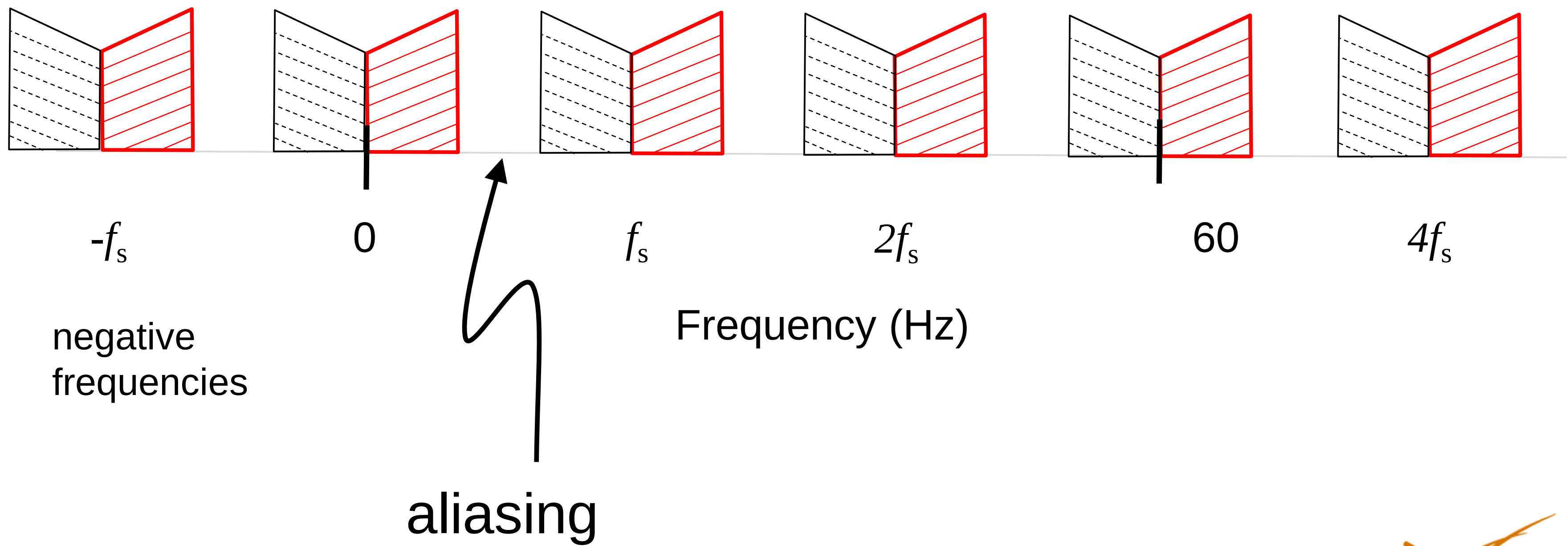
- Visualize the spectrum, assuming sampling faster than 2×BW:



But note that the sampling takes place before the DFT

So this spectrum is what the DFT operates on

Increase the sample rate:



That “Oh no!” feeling

- With “bandpass sampling” there IS an aliasing problem
- It exist all over the place
  - But the spectrum is usually relatively empty!

- But PMUs are not (as far as I know) undersampling

More on bandpass sampling

- Non-alias bandpass sampling is possible
- Depends [3] on
  - Sample rate
  - Width of band
  - Positioning of band

The numbers are a surprise!

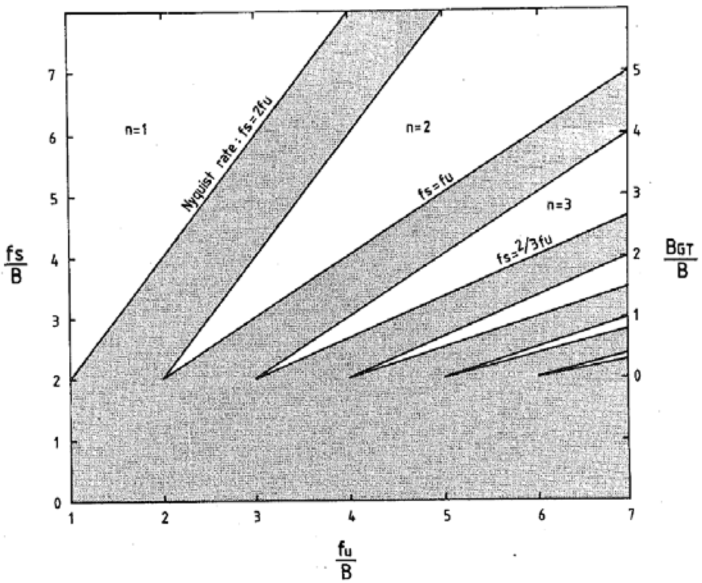
[3] Vaughan, R.G, Scott, N.L., White, D.R. "*The Theory of Bandpass sampling*"
IEEE Trans Signal Processing, Sept 1991, **39** (9), pp. 1973--1984

Turn the problem around: apply to PMU output

- Acceptable uniform sampling requires

$$\frac{2f_u}{n} \le f_s \le \frac{2f_L}{n-1}$$

n is the largest integer within f_u/B



- Suppose we (the user) want to look for oscillations in the power system
- *Now* we assume f_s is the reporting rate
- We need to find f_u for some hypothetical oscillation . Assume baseband sampling (n = 1)

$f_s = 30\ Hz$ acceptable for about a 12 Hz signal reconstruction

Just “ordinary” Nyquist

[3] Vaughan, R.G, Scott, N.L., White, D.R. “*The Theory of Bandpass sampling*” IEEE Trans Signal Processing, Sept 1991, **39** (9), pp. 1973--1984

What about that?

- User could make own filtering choices on measurement results

- Remember, each PMU measured result (on its own) is accurate

- If the user wants to look power system oscillation modes, he could get results up to about 12 Hz from a 30 per second rate

- Unless the PMU filters the results

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Change to the standard?

- Reconsider filter “recipe”
  - Nyquist not applied until reconstruction attempted
  - Need is therefore application-dependent
  - SSR relay example
- Standard should
  - not say “Nyquist frequency for reporting rate”
  - not be concerned with it – it is a user-only need
  - require no particular filtering applied to results of measurement
    - It reduces capability of PMU

Final Remarks

"Since the measuring device has been constructed by the observer, we have to remember that what we observe is not nature in itself, but nature exposed to our method of questioning" [4]

The PMU answers this question:
If this signal were a cosine wave, what would the amplitude, frequency and phase be?

But the signal may not be a cosine wave . . . and the *user* must decide what he wants measurement system to do

[4] Heisenberg, W. *Physics and Philosophy: The Revolution in Modern Science*, London: George Allen and Unwin, 1959.