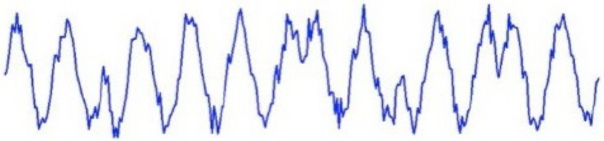
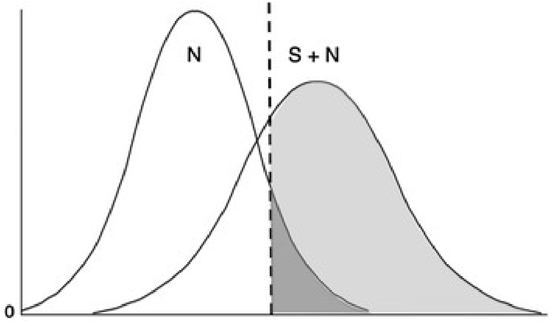
Aplicações de sinais

Tópicos sobre Detecção e Estimação … 2





Prof. Raul T. Rato

DEEC - 2021

Começando:

Apresentação para esta aula: (29 Abr)

Américo Alves 52418

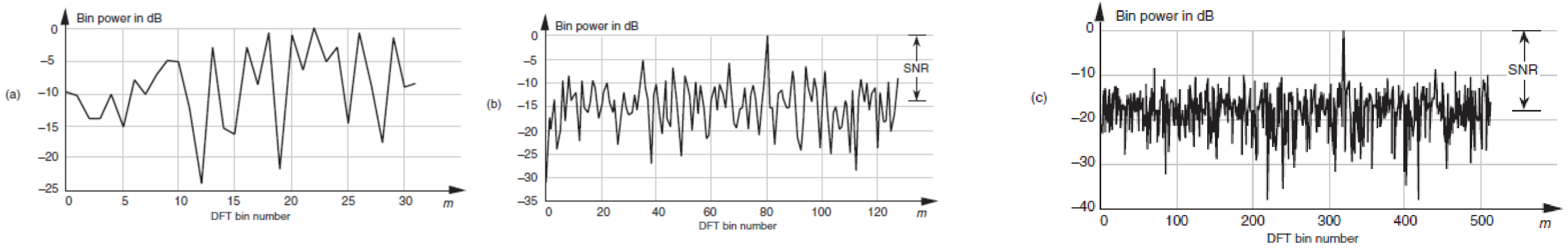
Descreva como o teorema de WK pode ser usado como auxiliar na busca por petróleo.

Nota: É interessante pesquisar sobre o MIT GAG

M.I.T. GEOPHYSICAL ANALYSIS GROUP (GAG)

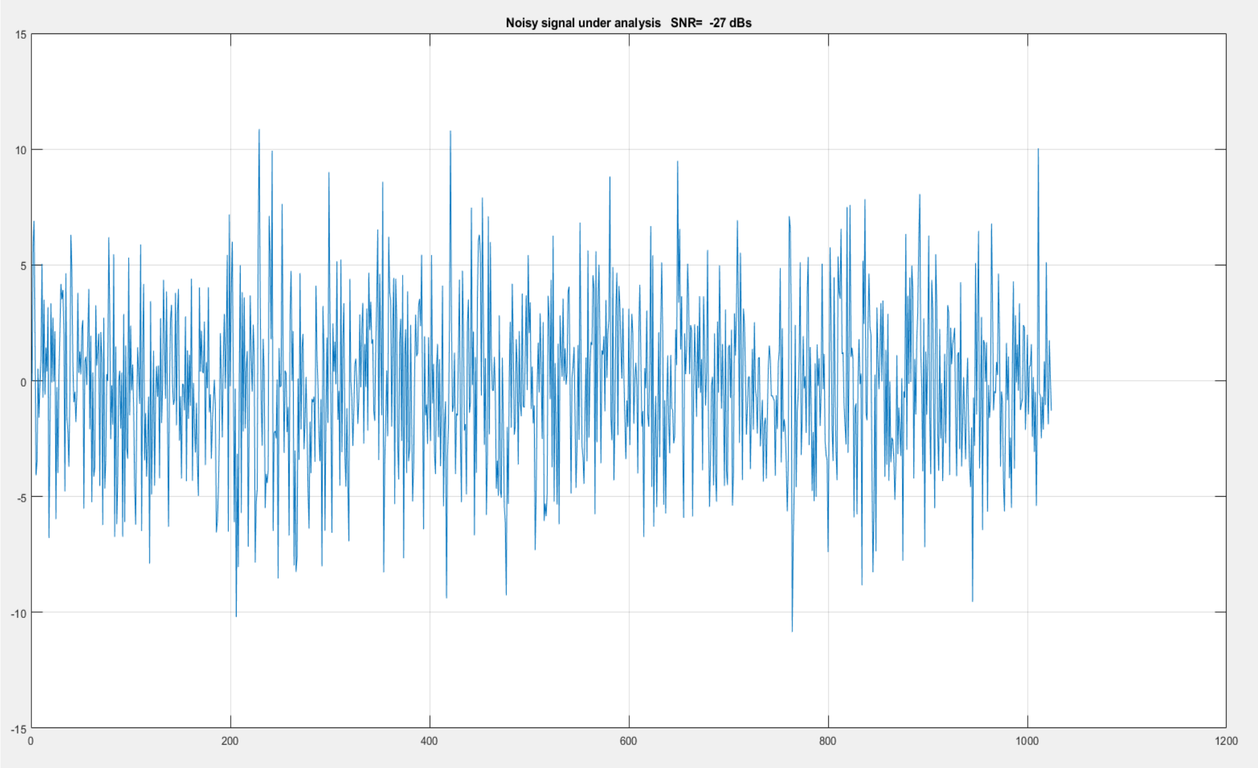
DFT PROCESSING GAIN

the DFT can *pull* signals out of background noise

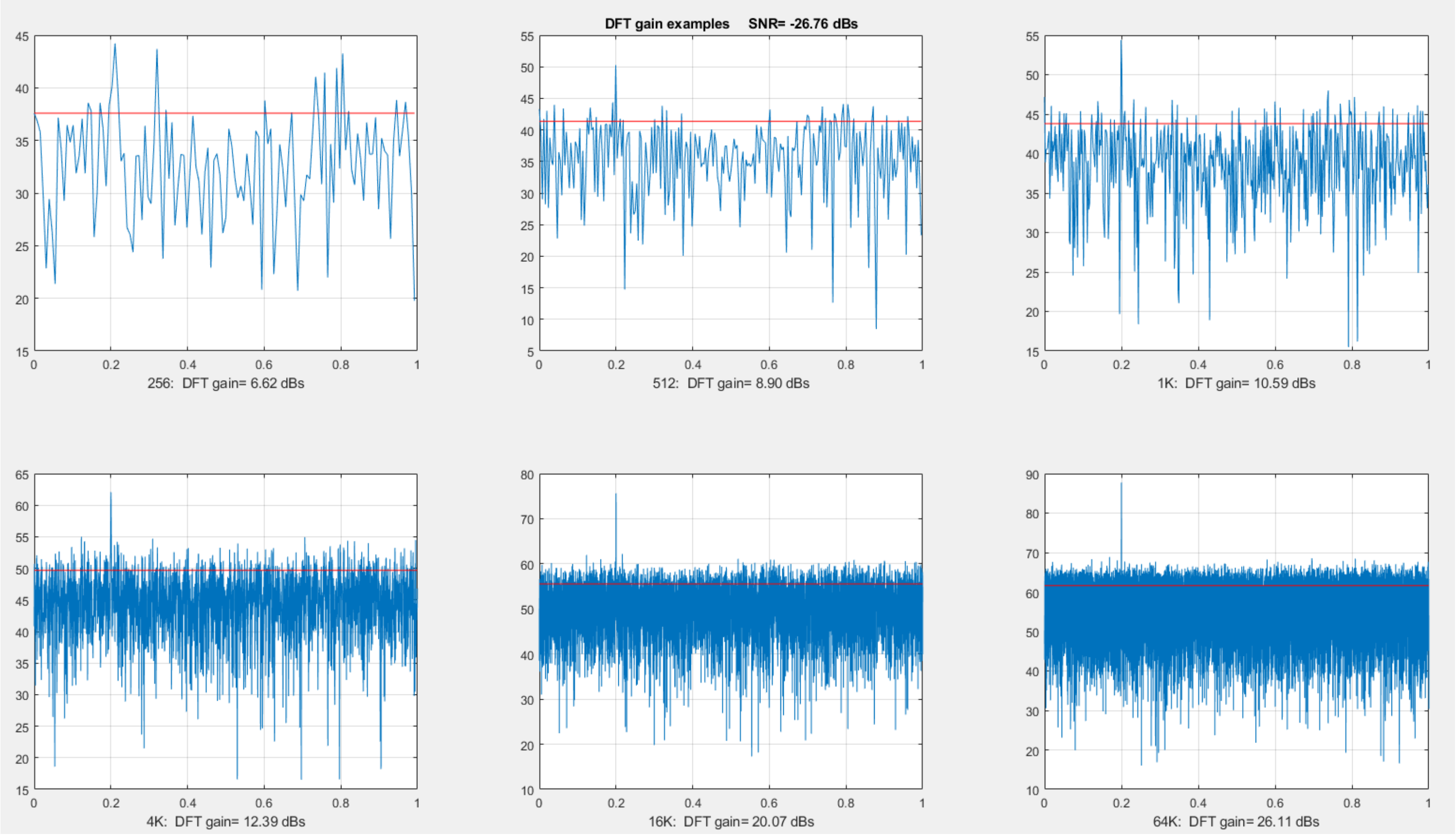


Single DFT processing gain: (a) $N = 64$; (b) $N = 256$; (c) $N = 1024$.

Tópicos sobre Detecção e Estimação … 2



Tem lá uma risca espectral? Mas ...



Notar que são (assintoticamente) +3dBs por cada duplicação

Tópicos sobre Detecção e Estimação … 2

Faz10DFTGain.m

```
clear
close all
clc
%
qNoisLevel= 3.3;
qN20= 1024*1024;
qN08=  256;
qN09=  512;
qN10= 1024;

qN12=  4*1024;
qN14= 16*1024;
qN16= 64*1024;

ktt= (0:(qN20-1))';

kNoise= qNoisLevel*randn(qN20,1);
kSinal= cos(0.2*pi*ktt);

qSNRdBs= 20*log10((kSinal'*kSinal)/(kNoise'*kNoise));
kR= kNoise+kSinal;

kSpct08= (20*log10(abs(fft(kR(1:qN08))))); qGain08= max(kSpct08)-mean(kSpct08)-std(kSpct08);
kSpct09= (20*log10(abs(fft(kR(1:qN09))))); qGain09= max(kSpct09)-mean(kSpct09)-std(kSpct09);
kSpct10= (20*log10(abs(fft(kR(1:qN10))))); qGain10= max(kSpct10)-mean(kSpct10)-std(kSpct10);
kSpct12= (20*log10(abs(fft(kR(1:qN12))))); qGain12= max(kSpct12)-mean(kSpct12)-std(kSpct12);
kSpct14= (20*log10(abs(fft(kR(1:qN14))))); qGain14= max(kSpct14)-mean(kSpct14)-std(kSpct14);
kSpct16= (20*log10(abs(fft(kR(1:qN16))))); qGain16= max(kSpct16)-mean(kSpct16)-std(kSpct16);

kff08= 2*(0:(1/(qN08)):(2*pi- (1/(qN08))))';
kff09= 2*(0:(1/(qN09)):(2*pi- (1/(qN09))))';
kff10= 2*(0:(1/(qN10)):(2*pi- (1/(qN10))))';

kff12= 2*(0:(1/(qN12)):(2*pi- (1/(qN12))))';
kff14= 2*(0:(1/(qN14)):(2*pi- (1/(qN14))))';
kff16= 2*(0:(1/(qN16)):(2*pi- (1/(qN16))))';

figure;

subplot(2,3,1); plot(kff08(1:(qN08/2)), kSpct08(1:(qN08/2)) );        %Only positive freqs
hold on; plot(kff08(1:(qN08/2)), (std(kSpct08)+ mean(kSpct08))* ones((qN08/2),1), 'r');
xlabel(sprintf('256:  DFT gain= %4.2f dBs', qGain08)); grid on;

subplot(2,3,2); plot(kff09(1:(qN09/2)), kSpct09(1:(qN09/2)) );        %Only positive freqs
hold on; plot(kff09(1:(qN09/2)), (std(kSpct09)+ mean(kSpct09))* ones((qN09/2),1), 'r');
xlabel(sprintf('512:  DFT gain= %4.2f dBs', qGain09)); grid on;
title(sprintf('DFT gain examples     SNR= %4.2f dBs', qSNRdBs));

subplot(2,3,3); plot(kff10(1:(qN10/2)), kSpct10(1:(qN10/2)) );        %Only positive freqs
hold on; plot(kff10(1:(qN10/2)), (std(kSpct10)+ mean(kSpct10))* ones((qN10/2),1), 'r');
xlabel(sprintf('1K:  DFT gain= %4.2f dBs', qGain10)); grid on;


subplot(2,3,4); plot(kff12(1:(qN12/2)), kSpct12(1:(qN12/2)) );        %Only positive freqs
hold on; plot(kff12(1:(qN12/2)), (std(kSpct12)+ mean(kSpct12))* ones((qN12/2),1), 'r');
xlabel(sprintf('4K:  DFT gain= %4.2f dBs', qGain12)); grid on;

subplot(2,3,5); plot(kff14(1:(qN14/2)), kSpct14(1:(qN14/2)) );        %Only positive freqs
hold on; plot(kff14(1:(qN14/2)), (std(kSpct14)+ mean(kSpct14))* ones((qN14/2),1), 'r');
xlabel(sprintf('16K:  DFT gain= %4.2f dBs', qGain14)); grid on;

subplot(2,3,6); plot(kff16(1:(qN16/2)), kSpct16(1:(qN16/2)) );        %Only positive freqs
hold on; plot(kff16(1:(qN16/2)), (std(kSpct16)+ mean(kSpct16))* ones((qN16/2),1), 'r');
xlabel(sprintf('64K:  DFT gain= %4.2f dBs', qGain16)); grid on;


figure;
plot(kR(1:qN10)); grid on
title(sprintf('Noisy signal under analysis    SNR= %4.f dBs', qSNRdBs));
```

To quantify the idea of DFT processing gain, we can define a signal-to-noise ratio (SNR) as the DFT's *output signal-power level* over the *average output noise-power level*. (In practice, of course, we like to have this ratio as large as possible.) For several reasons, it's hard to say what any given single DFT output SNR will be. That's because we can't exactly predict the energy in any given N samples of random noise. Also, if the input signal frequency is not at bin center, leakage will raise the effective background noise and reduce the DFT's output SNR. In addition, any window being used will have some effect on the leakage and, thus, on the output SNR. What we'll see is that the DFT's output SNR increases as N gets larger because a DFT bin's output noise standard deviation (*rms*) value is proportional to $\sqrt{N}$, and the DFT's output magnitude for the bin containing the signal tone is proportional to N.

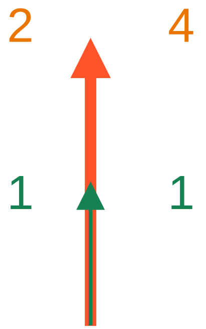
Importante. Notar que é uma afirmação de índole assintótica.

for real inputs, if $N > N'$, an N-point DFT's output SNR_N increases over the N'-point DFT SNR_N by the following relationship:

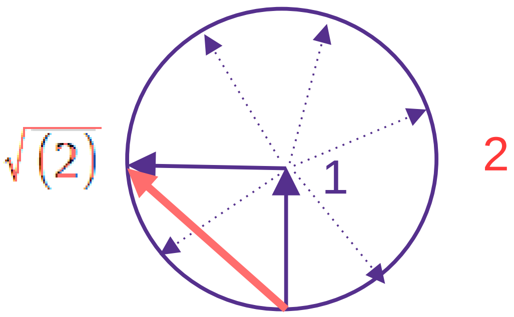
$$SNR_N = SNR_{N'} + 10\log_{10}\left(\frac{N}{N'}\right). \tag{3–33}$$

If we increase a DFT's size from N' to $N = 2N'$, from Eq. (3–33), the DFT's output SNR increases by 3 dB. So we say that a DFT's processing gain increases by 3 dB whenever N is doubled. Be aware that we may double a DFT's size and get a resultant processing gain of less than 3 dB in the presence of random noise; then again, we may gain slightly more than 3 dB. That's the nature of random noise.

O argumento vectorial (Existe uma demonstração muito mais rigorosa...)



Na componente sinal



Na componente ruído

$$\frac{4}{2}$$

No rácio SNR

MUITO IMPORTANTE

O argumento vectorial indicia-nos que
este efeito é geral e
não apenas particular à DFT

COHERENT AVERAGING

time-synchronous averaging

In the coherent averaging process
the key feature is the timing used in sampling the original signal

we collect multiple sets of signal-plus-noise samples, and we need the time phase of the signal in each set to be identical.

Let's say we collect 32 sets of 128 pulse-plus-noise samples of the form

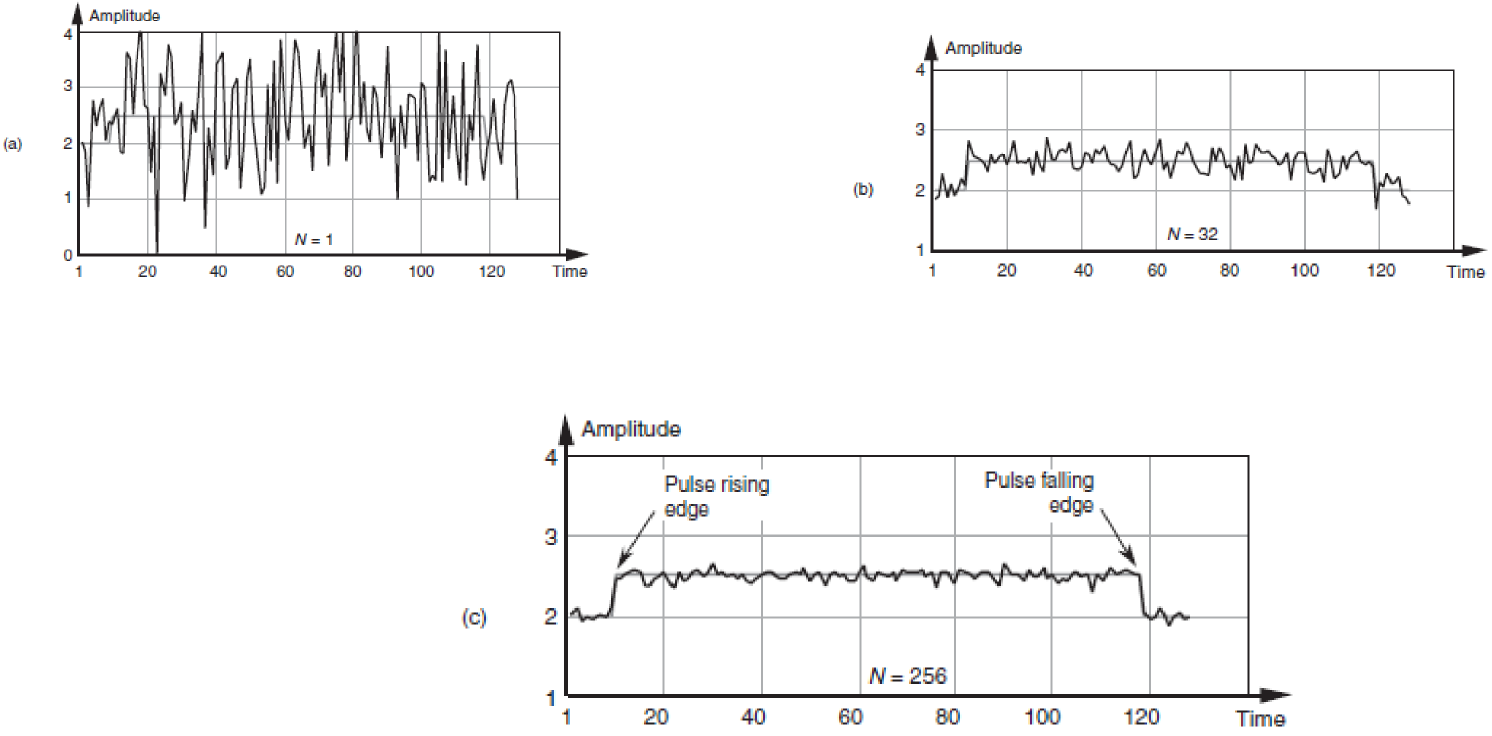
$$\begin{aligned}
&\text{Sample set}_1 = x_1(1), x_1(2), x_1(3), \ldots, x_1(128), \\
&\text{Sample set}_2 = x_2(1), x_2(2), x_2(3), \ldots, x_2(128), \\
&\text{Sample set}_3 = x_3(1), x_3(2), x_3(3), \ldots, x_3(128), \\
&\qquad\cdots \\
&\qquad\cdots \\
&\text{Sample set}_{32} = x_{32}(1), x_{32}(2), x_{32}(3), \ldots, x_{31}(128).
\end{aligned} \tag{11–4}$$

the signal measurement times must be synchronized, in some manner, with the beginning of the pulse, so that the pulse is in a constant time relationship with the first sample of each sample set.

$$
\begin{aligned}
x_{\text{ave}}(1) &= [x_1(1) + x_2(1) + x_3(1) + \ldots + x_{32}(1)]/32 \\
x_{\text{ave}}(2) &= [x_1(2) + x_2(2) + x_3(2) + \ldots + x_{32}(2)]/32 \\
x_{\text{ave}}(3) &= [x_1(3) + x_2(3) + x_3(3) + \ldots + x_{32}(3)]/32 \\
&\ldots \\
&\ldots \\
x_{\text{ave}}(128) &= [x_1(128) + x_2(128) + x_3(128) + \ldots + x_{32}(128)]/32 \quad .
\end{aligned}
\tag{11–5}
$$

É preciso garantir o sincronismo exacto – Minimizar o jitter

Tópicos sobre Detecção e Estimação … 2

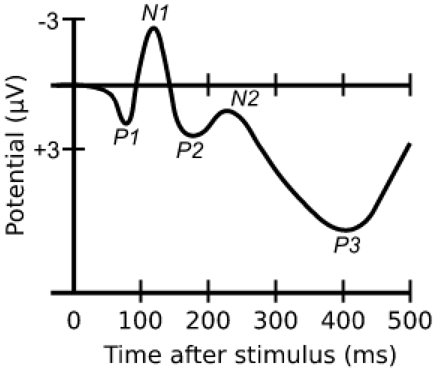


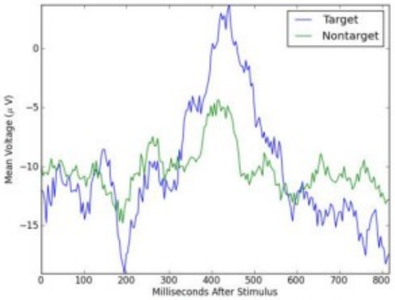
Signal pulse plus noise: (a) one sample set; (b) average of 32 sample sets; (c) average of 256 sample sets.

Through averaging, we can realize a signal-to-noise ratio improvement proportional to the square root of the number of signal samples averaged. In terms of signal-to-noise ratio measured in dB, we have a coherent averaging, or *integration*, gain of

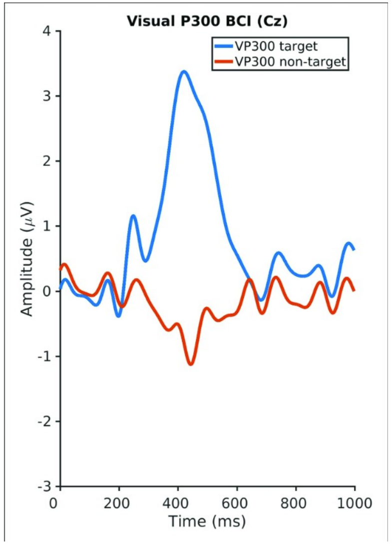
$$SNR_{\text{coh}}\ \text{gain(dB)} = 20 \cdot \log_{10}(SNR_{\text{coh}}) = 20 \cdot \log_{10}(\sqrt{N}) = 10 \cdot \log_{10}(N). \tag{11–14}$$

Isto é muito importante para estimar sinais repetitivos afogados em ruído

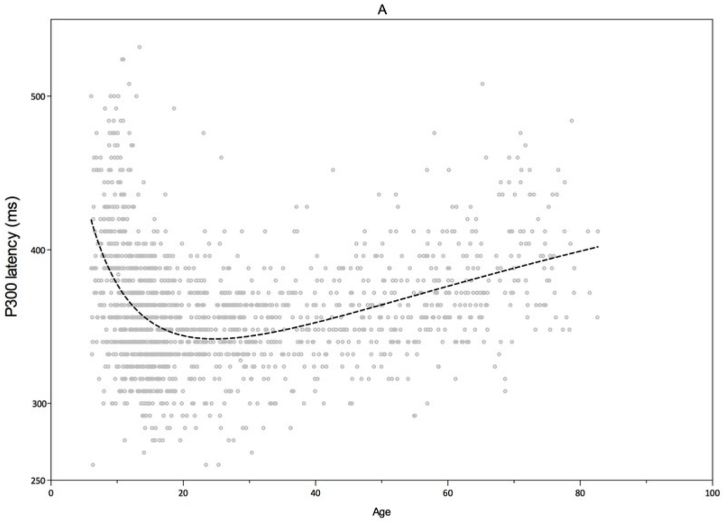


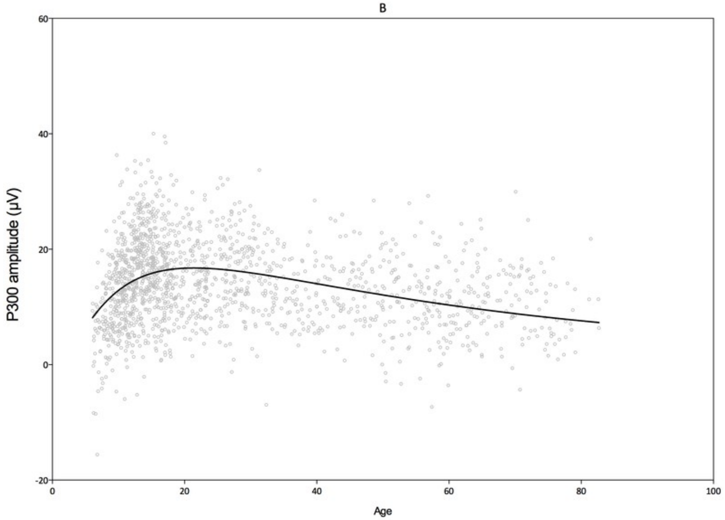


A typical P300 wave

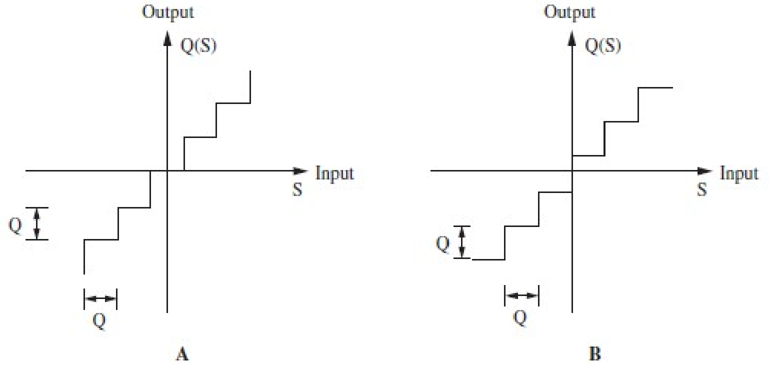


Tópicos sobre Detecção e Estimação … 2





Rik van Dinteren, Martijn Arns, Marijtje L. A. Jongsma, Roy P. C. Kessels -
P300 Development across the Lifespan: A Systematic Review and Meta-Analysis. 2014



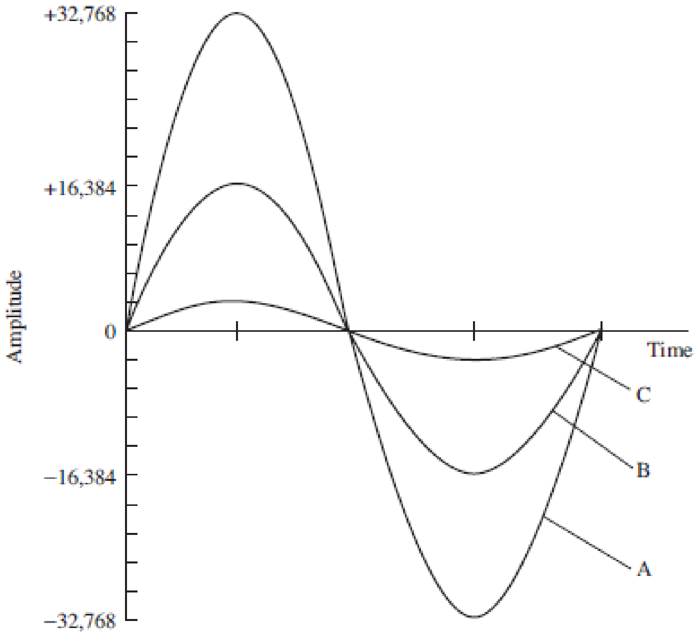
Signals can be quantized in one of two ways. A. A midtread quantizer. B. A midrise quantizer. Q (or 1 LSB) is the quantizer step size.

Ruído de quantificação em função do número de bits no ADC:

$$S/E\,\mathrm{dB} = 10\log\left[\frac{3}{2}(2^{2n})\right]$$
$$= 20\log\left[\left(\frac{3}{2}\right)^{1/2}(2^{n})\right]$$
$$= 6.02n + 1.76\,\mathrm{dB}$$

Esta fórmula para o ruído de quantificação em função do número de bits é válida quando a excursão do sinal for completamente ampla, de máximo a mínimo do ADC

Principles of Digital Audio 6Ed - Pohlmann -2011



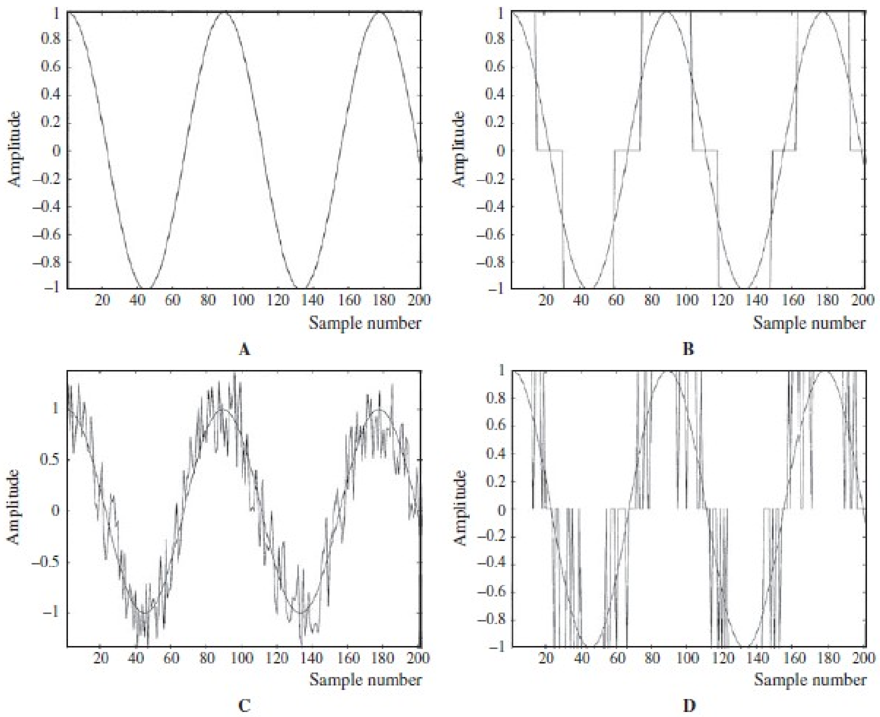
The percentage of quantization error increases as the signal level decreases. Full-scale waveform A has relatively low error (16-bit resolution). Half-scale waveform B has higher error (effectively 15-bit resolution). Low-amplitude waveform C has very high error.

Dither

With large-amplitude complex signals, there is little correlation between the signal and the quantization error; thus, the error is random and perceptually similar to analog white noise. With low-level signals, the character of the error changes as it becomes correlated to the signal, and potentially audible distortion results.

Dither does not mask quantization error; rather, it allows the digital system to encode amplitudes smaller than the least significant bit.

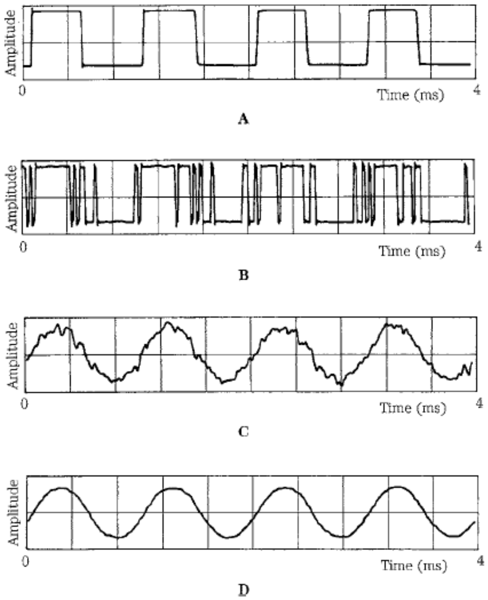
Tópicos sobre Detecção e Estimação … 2



Dither is used to alleviate the effects of quantization error. A. An undithered input sine wave signal with amplitude of two LSBs. B. Quantization results in a coarse coding over three levels. C. Dither is added to the input sine wave signal. D. Quantization yields a PWM waveform that codes information below the LSB.

Tópicos sobre Detecção e Estimação … 2

John Vanderkooy and Stanley Lipshitz demonstrated the remarkable benefit of dither with a 1-kHz sine wave with a peak-to-peak amplitude of about 1 LSB, as shown in Fig. 2.10. Without dither, a square wave is output (Fig. 2.10A). When wideband Gaussian dither with an rms amplitude of about 1/3 LSB is added to the original signal before quantization, a pulse-width-modulated (PWM) waveform results (Fig. 2.10B).



Dither permits encoding of information below the least significant bit. A. Quantizing a 1-kHz sine wave with peak-to-peak amplitude of 1 LSB without dither produces a square wave. B. Dither of 1/3 LSB rms amplitude is added to the sine wave before quantization, resulting in PWM modulation. C. Modulation conveys the encoded sine wave information, as can be seen after 32 averagings. D. The encoded sine wave information is more apparent after 960 averagings. (*Vanderkooy and Lipshitz, 1984*)

Notar que a frequência de amostragem tem de ser bastante alta relativamente à da sinusoide

Está relacionado com
a ressonância estocástica

Ressonância estocástica

Stochastic resonance (SR) is a phenomenon in which a signal that is normally too weak to be detected by a sensor, can be boosted by adding white noise to the signal.

The frequencies in the white noise corresponding to the original signal's frequencies will “resonate” with each other, amplifying the original signal while not amplifying the rest of the white noise – thereby increasing the signal-to-noise ratio.

Further, the added white noise can be enough to be detectable by the sensor, which can then filter it out to effectively detect the original, previously undetectable signal.

This phenomenon of boosting undetectable signals by resonating with added white noise extends to many other systems – whether electromagnetic, physical or biological – and is an active area of research.

SR occurs if the signal-to-noise ratio of a nonlinear system or device increases for moderate values of noise intensity.

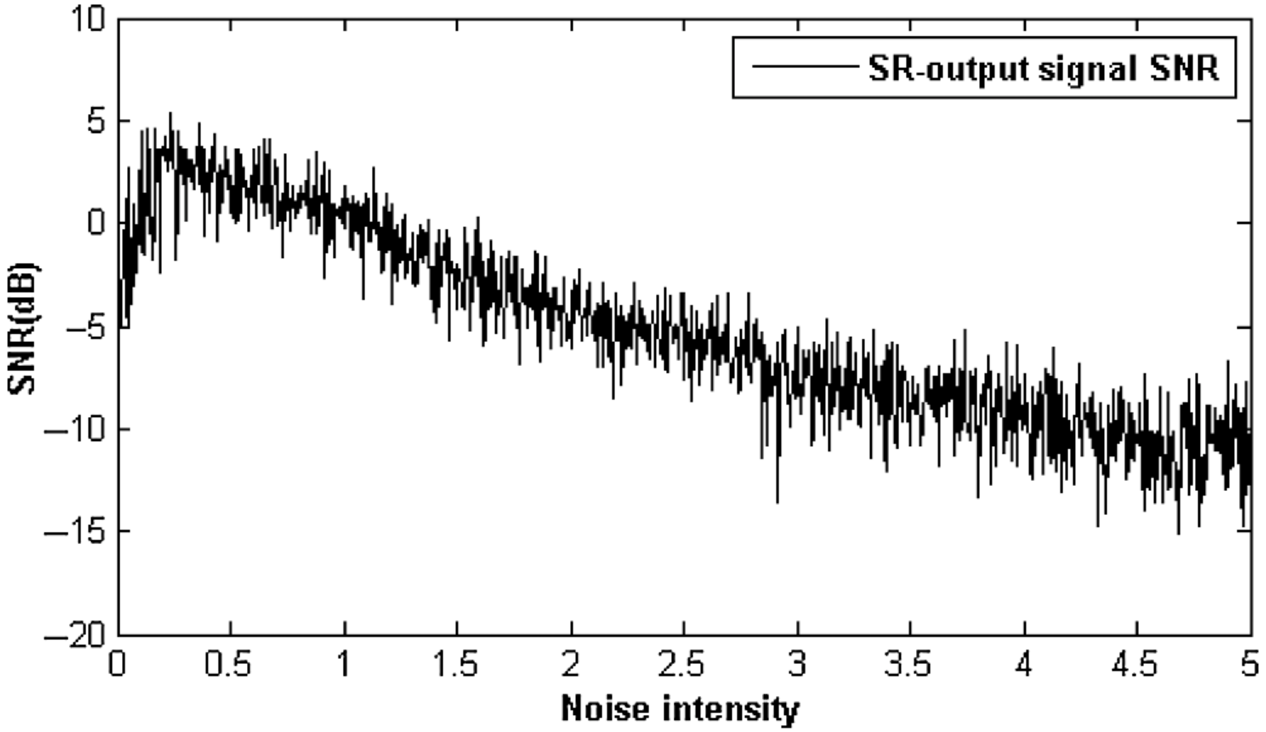
It often occurs in bistable systems or in systems with a sensory threshold and when the input signal to the system is "sub-threshold."

- For lower noise intensities, the signal does not cause the device to cross threshold, so little signal is passed through it.

- For large noise intensities, the output is dominated by the noise, also leading to a low signal-to-noise ratio.

- For moderate intensities, the noise allows the signal to reach threshold, but the noise intensity is not so large as to swamp it.

Thus, a plot of signal-to-noise ratio as a function of noise intensity contains a peak.



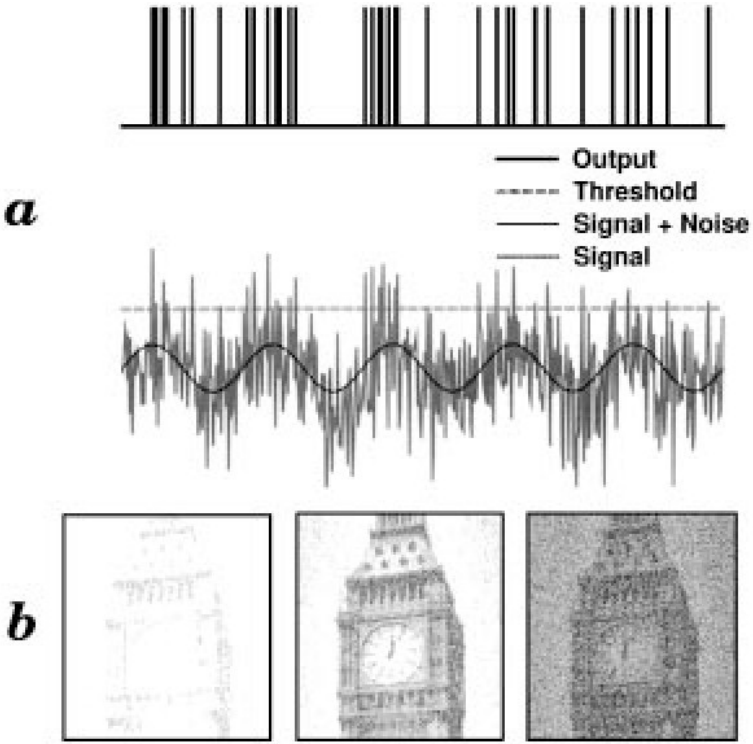
SR-output signal SNR
SNR(dB)
Noise intensity
10
5
0
−5
−10
−15
−20
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
5

Terminando:

Apresentação para a próxima aula: (4 Mai)

Procure e apresente um artigo de investigação em que se lide com a ressonância estocástica.

Avançado: Se for num contexto de probabilidades variáveis no tempo, ainda melhor.



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