

Electronics – 96032



POLITECNICO DI MILANO



Lock-in Amplifiers

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Slides are supplementary material and are NOT a replacement for textbooks and/or lecture notes



Purpose of the lesson

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- To discuss how real signal recovery systems based on amplitude modulation are designed
- To highlight a few parameters and performance



What LIAs are

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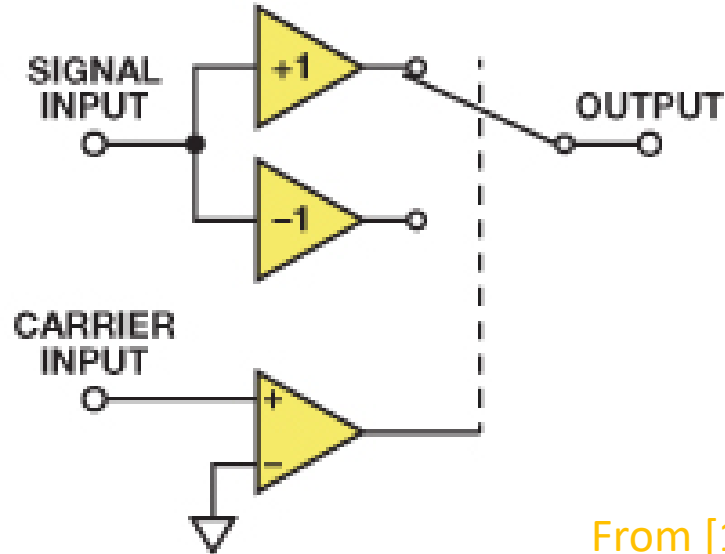
- LIAs are powerful tools that make use of a PSD to recover weak modulated signals buried in noise
- Several modifications are made to the basic scheme in order to reach high performance



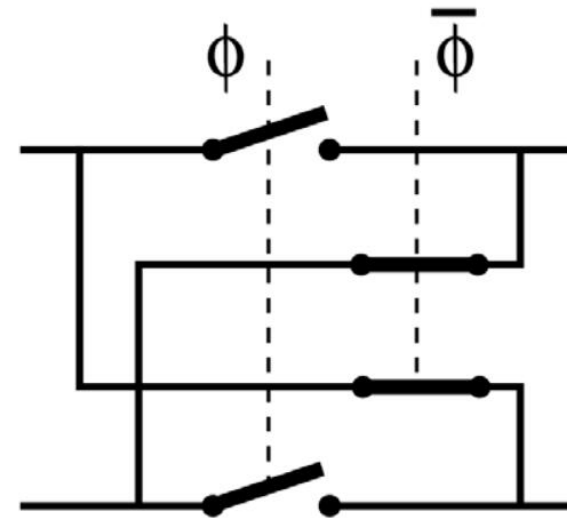
Switching demodulators

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- Analog multipliers are expensive and prone to nonlinearity errors
- Modulators/mixers are rather used, where the signal is multiplied by the **sign** of the carrier (i.e., by a square wave at f_r)



From [1]





Advantages

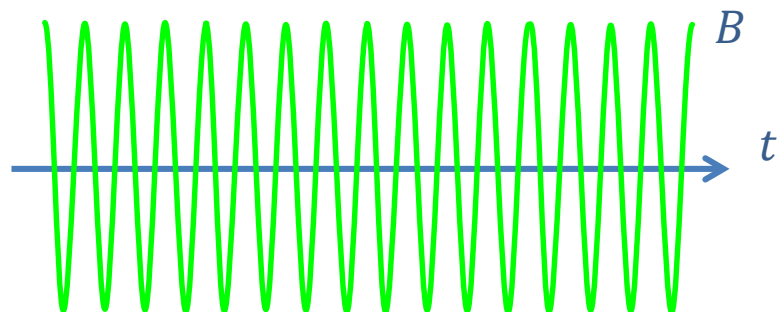
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- Any noise or amplitude variation on the reference can be ignored
- The noise is mainly due to the switches and is usually lower than in multipliers
- Easier to design and manufacture



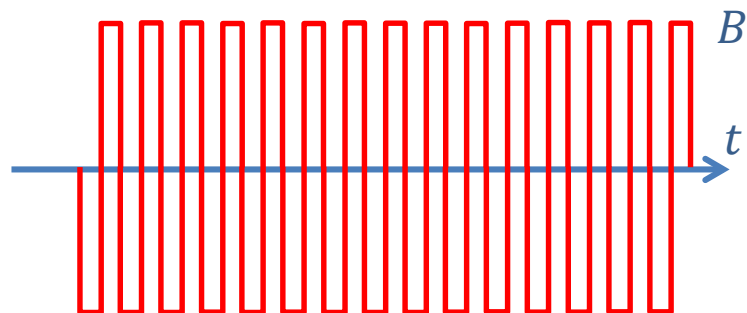
Reference functions – time domain

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Sinusoidal case:

$$w_R(t) = B \cos(\omega_r t)$$



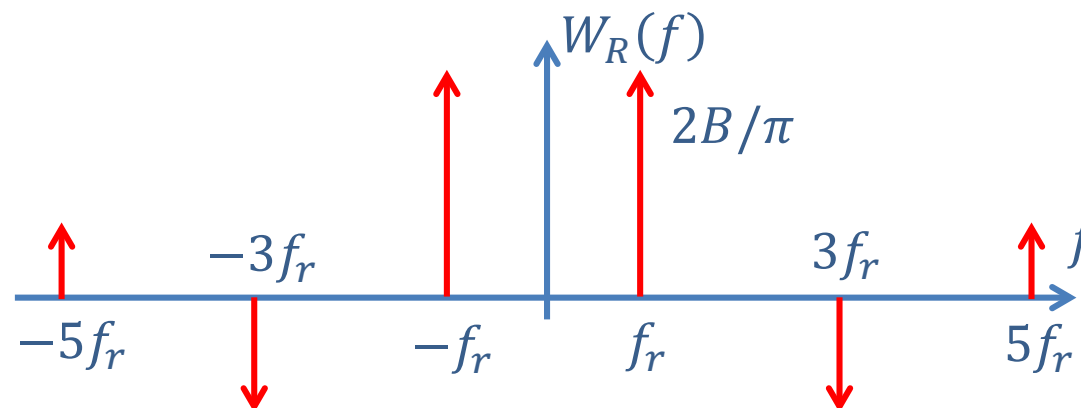
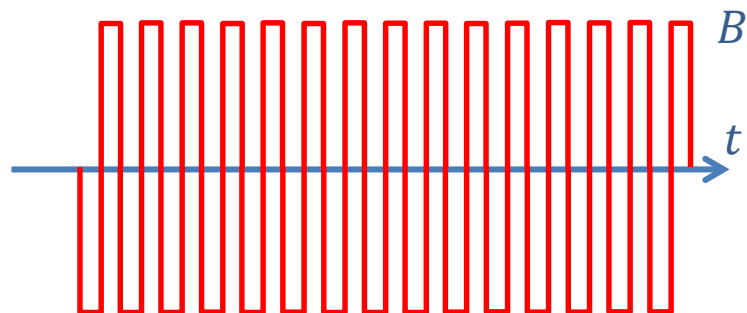
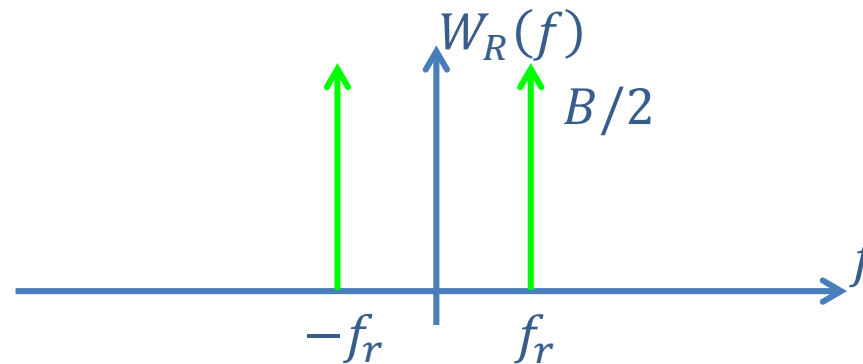
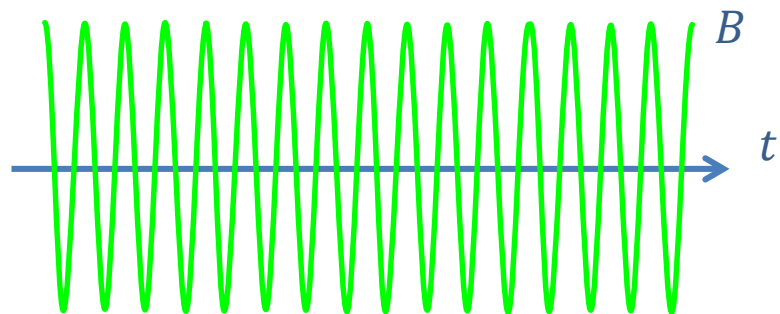
Square-wave case:

$$w_R(t) = \frac{4B}{\pi} \sum_k \frac{(-1)^k}{2k+1} \cos((2k+1)\omega_r t)$$



Reference functions – frequency domain

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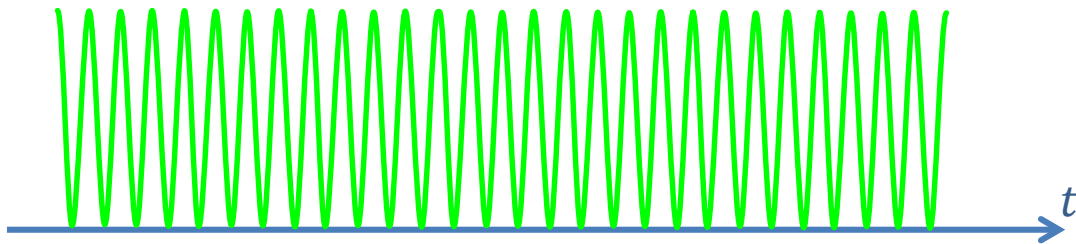


$$W_R(f) = \frac{2B}{\pi} \sum_k \frac{(-1)^k}{2k+1} (\delta(f - (2k+1)f_r) + \delta(f + (2k+1)f_r))$$



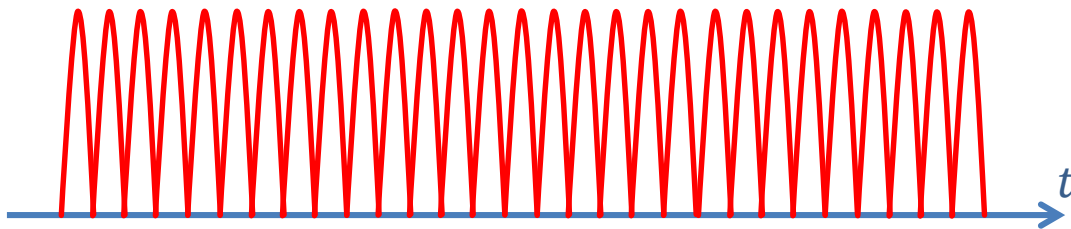
Mixer output signals

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Sinusoidal
demodulation

$$d(t) = AB \cos^2(\omega_r t)$$



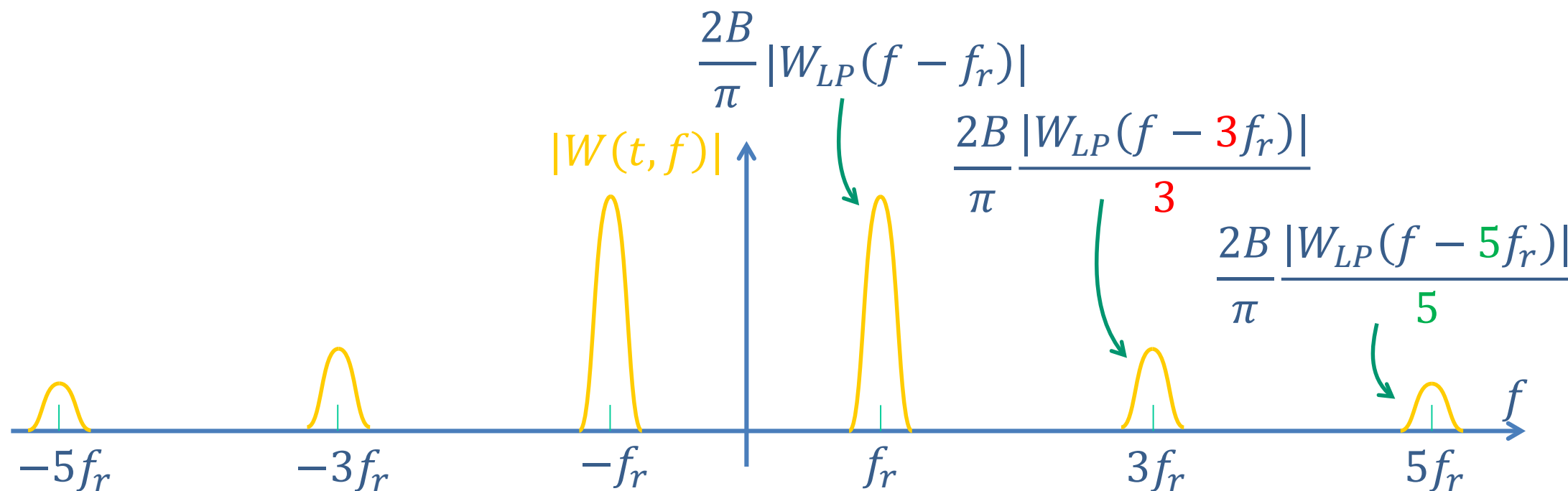
Square-wave
demodulation

$$d(t) = AB | \cos(\omega_r t) |$$



Switching PSD spectral response

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$$W(t, f) = W_R(f) * W_{LP}(f)$$



$$w_R(t) = \sum_k 2B_k \cos(k\omega_r t) \Leftrightarrow$$

$$W_R(f) = \sum_k B_k (\delta(f - kf_r) + \delta(f + kf_r))$$

$$K_{w_R w_R}(\tau) = \sum_k 2B_k^2 \cos(k\omega_r \tau) \Leftrightarrow$$

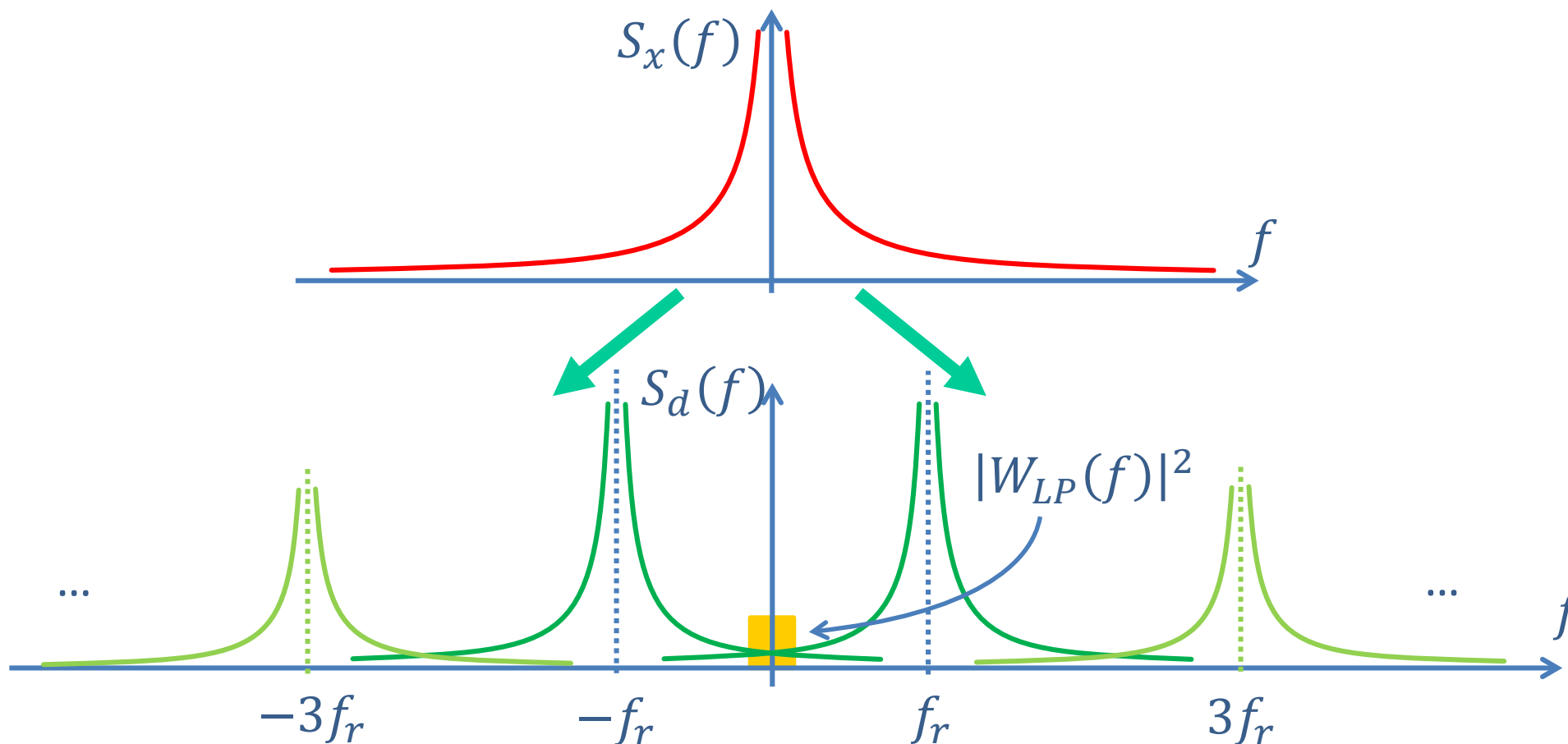
$$S_{w_R}(f) = \sum_k B_k^2 (\delta(f - kf_r) + \delta(f + kf_r))$$

B_k = amplitude of delta functions (Fourier coefficients)



Ex: input LF noise

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$$S_d(f) = S_x(f) * S_{W_R}(f) = \sum_k B_k^2 (S_x(f - kf_r) + S_x(f + kf_r))$$



The output filter collects noise from all replicas:

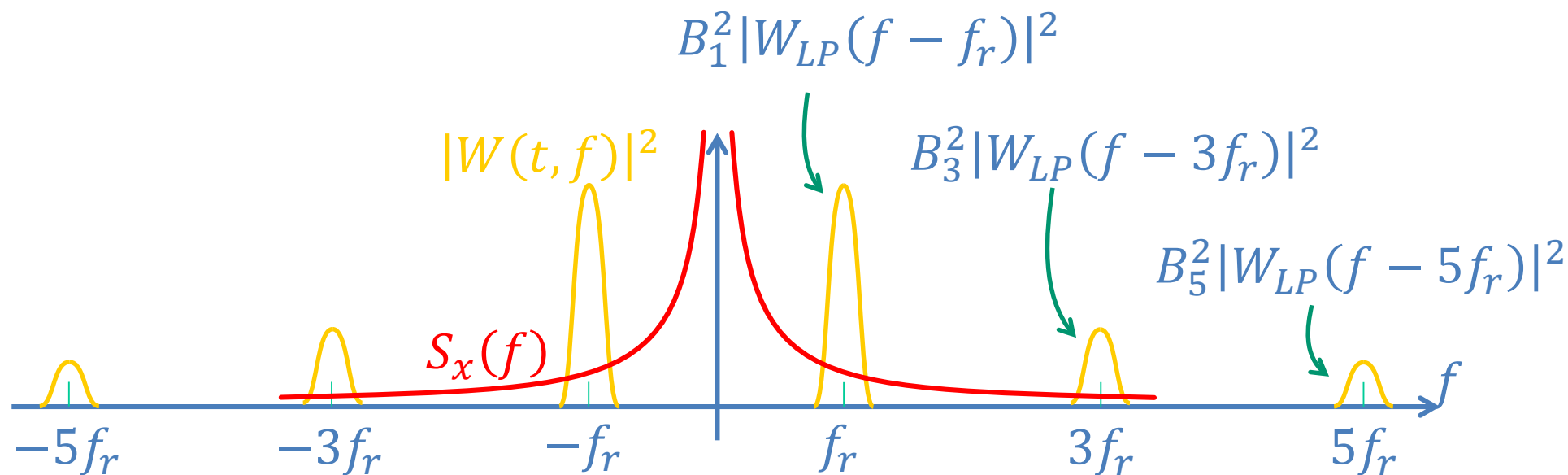
$$\begin{aligned}\overline{n_y^2} &= \int S_d(f) |W_{LP}(t, f)|^2 df \approx S_d(0) 2BW_n \\ &= 2BW_n \sum_k B_k^2 (S_x(-kf_r) + S_x(kf_r)) = 4BW_n \sum_k B_k^2 S_x(kf_r) \\ &= 4BW_n B_1^2 S_x(f_r) \sum_k \frac{B_k^2}{B_1^2} \frac{S_x(kf_r)}{S_x(f_r)}\end{aligned}$$

Extra noise collected by the harmonics



Another look at noise collection

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The output noise is controlled by the spectral response of the PSD



S/N ratio (square wave)

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- We consider a constant (sin-modulated) signal $x(t) = A \cos(\omega_r t)$
- Output signal and noise are

$$y(t) = A \frac{2B}{\pi} \quad \overline{n_y^2} = 4BW_n \left(\frac{2B}{\pi} \right)^2 \sum_k \frac{S_x((2k+1)f_r)}{(2k+1)^2}$$

$$\frac{S}{N} = \frac{A}{\sqrt{4BW_n \sum_k \frac{S_x((2k+1)f_r)}{(2k+1)^2}}} = \frac{A}{\sqrt{4BW_n S_x(f_r) \sum_k \frac{S_x((2k+1)f_r)}{S_x(f_r) (2k+1)^2}}}$$



- If we are modulating beyond the noise corner frequency, $S_x((2k + 1)f_r)$ is white (constant) and

$$\sum_k \frac{1}{(2k + 1)^2} = \frac{\pi^2}{8} \approx 1.2337$$

$$\left(\frac{S}{N}\right)_{\text{sqd}} = \frac{1}{\sqrt{1.2337}} \left(\frac{S}{N}\right)_{\text{sin}} = \frac{1}{1.11} \left(\frac{S}{N}\right)_{\text{sin}}$$

- S/N is (slightly) degraded because of the extra noise collected at the higher harmonics, where there is no signal



- If instead there is a pure $1/f$ input noise spectrum, we have

$$S_x((2k + 1)f_r) = \frac{Const}{(2k+1)f_r} \text{ and the extra noise becomes}$$

$$\sum_k \frac{1}{(2k + 1)^3} \approx 1.0518$$

$$\left(\frac{S}{N}\right)_{\text{sqd}} = \frac{1}{\sqrt{1.0518}} \left(\frac{S}{N}\right)_{\text{sin}} = \frac{1}{1.0256} \left(\frac{S}{N}\right)_{\text{sin}}$$



- If the signal is also modulated by a square wave (full-square demodulation), we get (white noise case)

$$y(t) = AB = \frac{\pi}{2} (y(t))_{\sin} \quad \overline{n_y^2} = \frac{\pi^2}{8} (\overline{n_y^2})_{\sin}$$

$$\left(\frac{S}{N}\right)_{\text{fsq}} = \frac{\pi}{2} \frac{\sqrt{8}}{\pi} \left(\frac{S}{N}\right)_{\sin} \approx 1.41 \left(\frac{S}{N}\right)_{\sin}$$

- Note that in the sinusoidal case, the signal is the first harmonic of the square wave

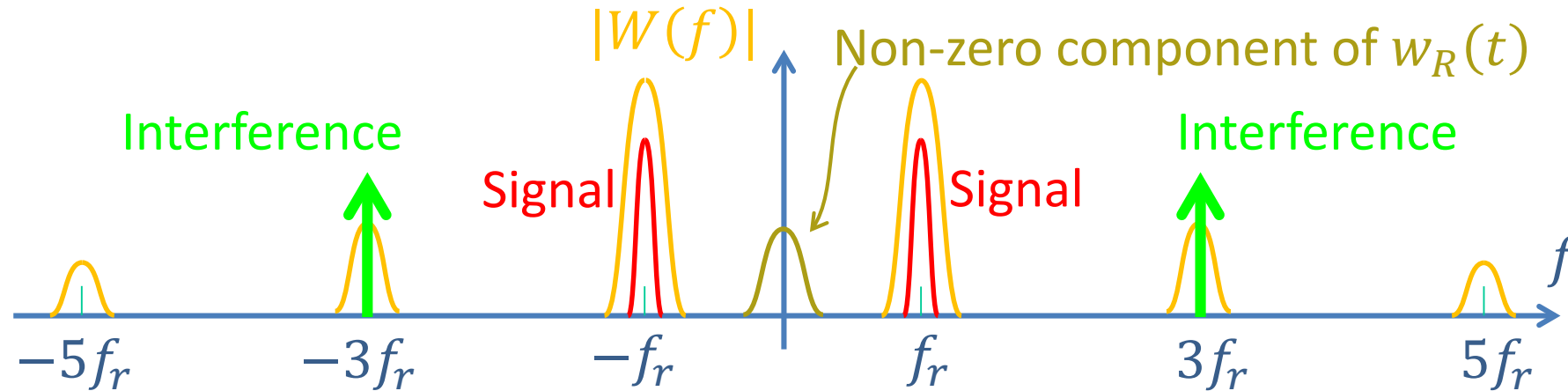


- For square-wave signal and sinusoidal demodulation we would get

$$y(t) = \frac{4AB}{\pi} = \frac{4}{\pi} (y(t))_{\sin} \quad \overline{n_y^2} = (\overline{n_y^2})_{\sin}$$

$$\left(\frac{S}{N}\right)_{\text{sqs}} = \frac{4}{\pi} \left(\frac{S}{N}\right)_{\sin} \approx 1.27 \left(\frac{S}{N}\right)_{\sin}$$

- Note that the sinusoidal demodulation only detects the first harmonic of the square wave, with amplitude $4A/\pi$

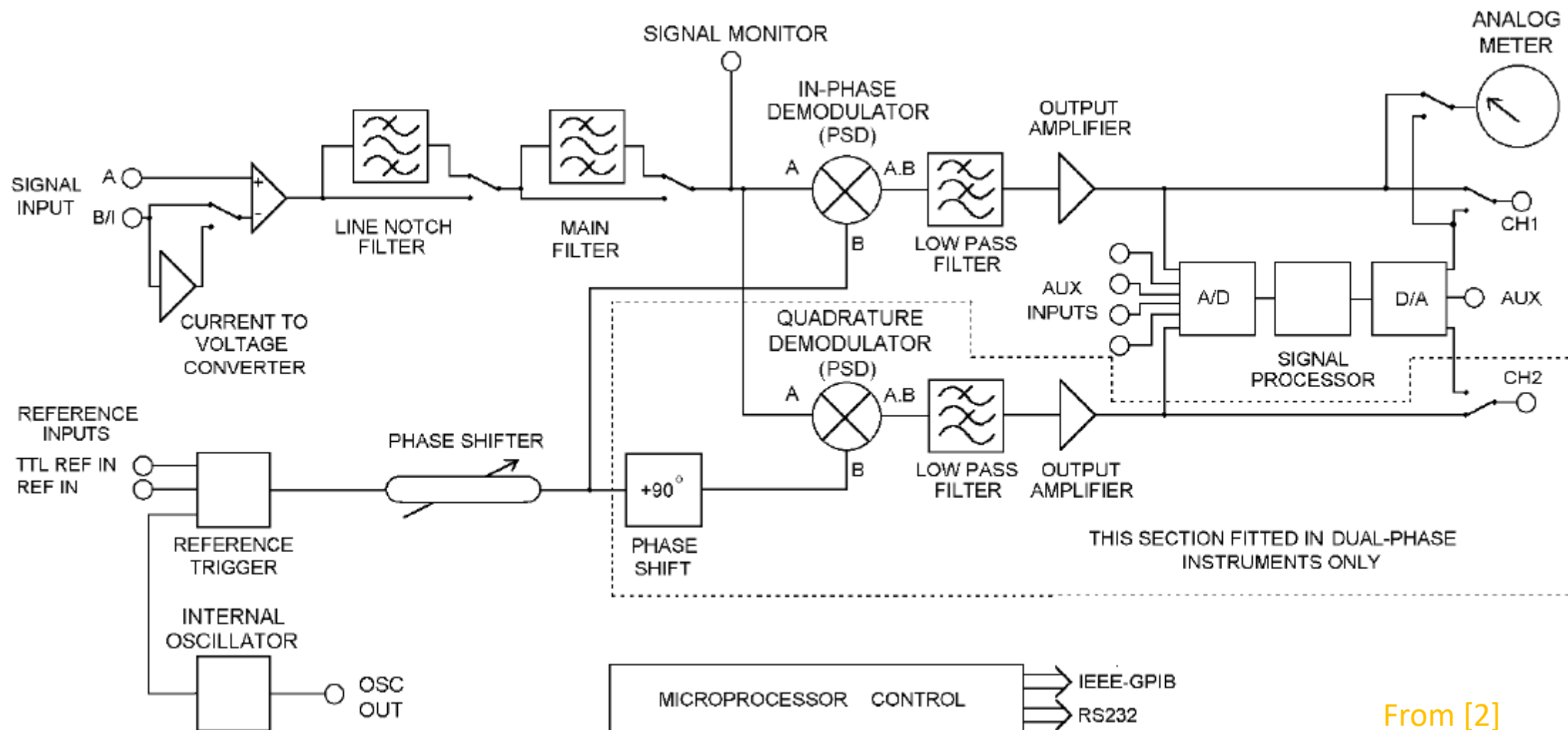


- If $\langle w_R(t) \rangle \neq 0$ the lock-in will collect low-frequency noise, wasting all efforts (also true for sinusoidal modulation)
- The output is sensitive to localized interferences at the harmonics of $f_r \Rightarrow$ keep f_r away from multiples of 50 (or 60) Hz



Actual structure

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From [2]



- Notch filter: removes 50/60 Hz and/or 2nd harmonics from signal (check Q value)
- Main filter: either LP, BP or notch
 - BP is usually employed (cleans up the signal improving dynamic reserve – see later)
 - LP can be used for low f_r
 - Flat response (no filter) can be selected for non-demanding applications
 - Always check the roll-off of the filter



- Reference trigger: generates a trigger signal (e.g. sqw) synchronous with the reference (or at $2f_r$)
- Phase shifter: adjusts the phase of the reference to track the signal
- PSD: usually a square-wave mixer. More refined implementations (e.g. suppressing 3rd harmonic response) may be employed
- Two channels working in quadrature are always employed



- Low-pass filter: either first- or second-order filters, usually specified via the time constant
 - 1st order may be not good enough in rejecting aliases
 - 2nd order may give problems if LIA is used within a feedback loop
- Output amplifier: provides gain increasing the sensitivity. Since it is DC-coupled, its drift and noise must be controlled



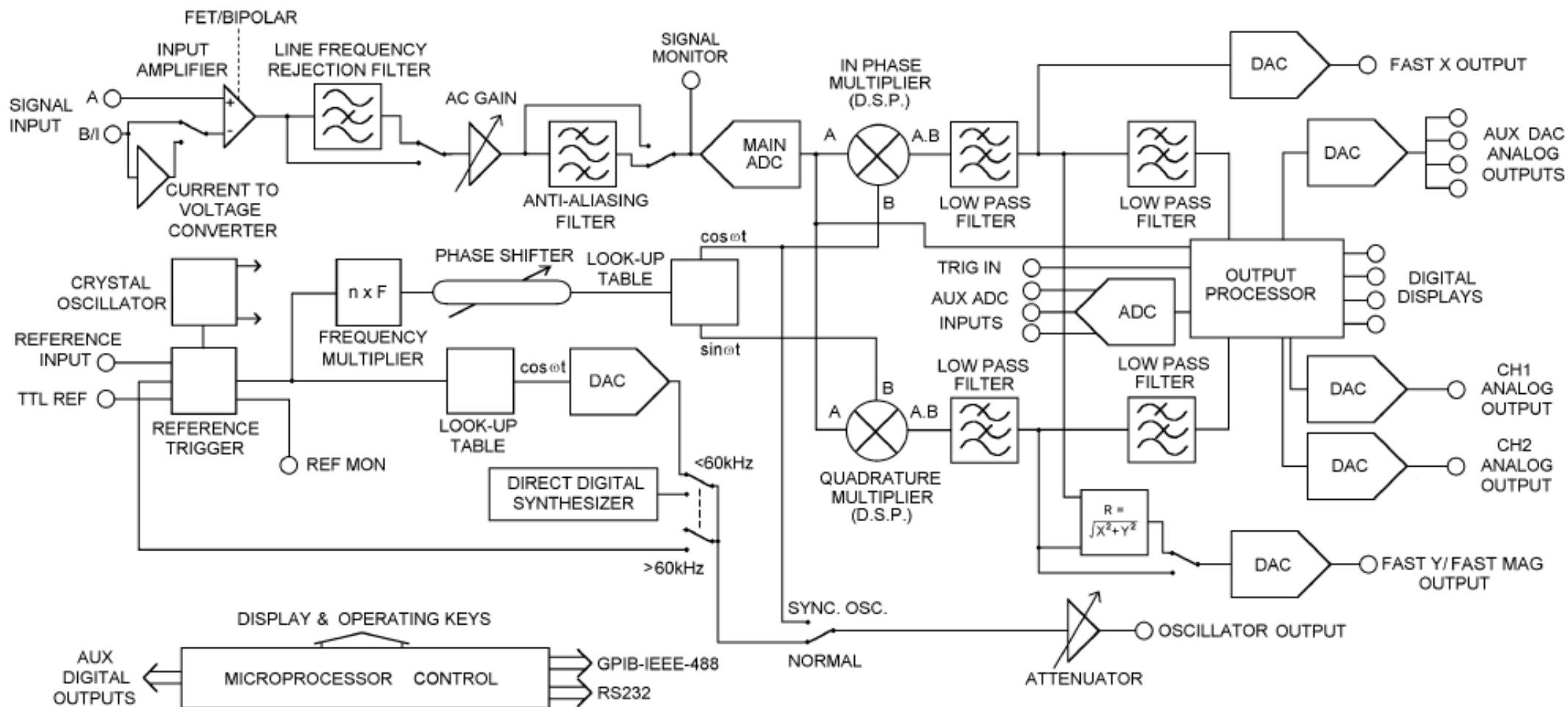
- Frequency range: typically from a fraction of Hz to a few MHz (200 MHz for RF LIAs)
- Output time constants: ms to thousands of s
- Dynamic reserve: is the ratio of the largest “tolerable” noise signal to the full scale signal, expressed in dB. Typical values are ≤ 60 dB



More on the dynamic reserve

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- “tolerable” noise refers to a specified precision in the measurement of the signal
- DR depends on the frequency of the noise. The quoted value is the **maximum** value, achievable when noise is away from f_r
- DR is eventually limited by PSD non-linearity and overload
- More information in [3]



The PSD is implemented in digital circuitry

From [4]



- Reference signals are pre-computed with high precision \Rightarrow better PSD performance (offset, harmonics,...). DR of 100 dB can be achieved
- Much lower f_r (mHz) are allowed
- More flexibility in the (digital) LPF design (higher orders, longer time constants,...)
- No output amplifier \Rightarrow better stability



What is left for analog LIAs?

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- Demodulation at high frequencies (MHz), since f_r is limited to a few MHz for digital LIAs
- Applications requiring short time constants at mid-range frequencies (≈ 100 kHz)
- Feedback loops employing LIAs
- More discussion on analog and digital LIAs in [5]



A comparison chart

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Make/Model	Features	Price Range	Remarks
SRS Model SR830	<ul style="list-style-type: none">1 mHz to 102.4 kHz frequency range256 kHz front-end sampling rate> 100 dB dynamic reserve (5 ppm/C stability)Auto gain, phase, reverse, and offset0.01 deg phase resolutionTime constant from 10us to 30 ksGPIB, and RS-232 interfaces	~ \$5,200	DSP Lock-in Amplifier with 41/2-digit LED display. Power 40 W 100/120/220/240 VAC, 50/60Hz
SRS Model SR850	<ul style="list-style-type: none">1 mHz to 102.4 kHz frequency rangeCRT display, and built-in data analysis64,000 point data bufferChart recording, numeric display, polar plotsSmoothing, curve fitting, and statisticsDirect printing, and plotting0.001 deg phase resolution>100 dB dynamic reserveGPIB, and RS-232 interfaces	~ \$8,700	DSP Lock-in Amplifier. Power 60W 100/120/220/240 VAC, 50/60Hz
SRS Model SR844	<ul style="list-style-type: none">25 kHz to 200 MHz frequency range80 dB dynamic reserve (5 ppm stability)Auto gain, phase, reverse, and offsetInternal or external referenceTime constant from 100us to 30 ksNo Time Constant modeGPIB, and RS-232 interfaces	~ \$9,200	RF Lock-in Amplifier For high frequency applications.
Perkin Elmer/ EG&G Model 7265	<ul style="list-style-type: none">0.001 Hz to 250 kHz frequency rangeVoltage and current mode inputs10 us to 100 ks output time constantsQuartz crystal stabilized internal oscillatorHarmonic measurements to 65,536FDual reference, Dual harmonic, and Virtual Reference modesSpectral display modes	~\$5,300	Dual Phase DSP Lock-in Amplifier
Perkin Elmer/ EG&G Model 7280	<ul style="list-style-type: none">0.5 Hz to 2 MHz frequency rangeVoltage and current mode inputs7.5 MHz main ADC sampling rate1 us to 100 ks output time constantsQuartz crystal stabilized internal oscillatorHarmonic measurements to 32FDual reference, Dual harmonic, and Virtual Reference modesSpectral display mode	~\$12,500	Dual Phase wide bandwidth DSP Lock-in Amplifier. Power 200 VA max 100/120/220/240 VAC, 50/60Hz

From [6]



1. http://www.analog.com/library/analogdialogue/archives/47-06/multipliers_modulators.html
2. <http://cpm.uncc.edu/sites/cpm.uncc.edu/files/media/tn1002.pdf>
3. <http://www.thinksrs.com/downloads/PDFs/ApplicationNotes/AboutLIAs.pdf>
4. <http://mrflip.com/papers/LIA/LIAProgram/Support/References/AmetekSR-TechNote1003.pdf>
5. http://home.deib.polimi.it/cova/elet/lezioni/SSN08d_Filters-BPF4.pdf
6. <http://www.vinkarola.com/pdf/Lock-InAmplifiersComparison.pdf>