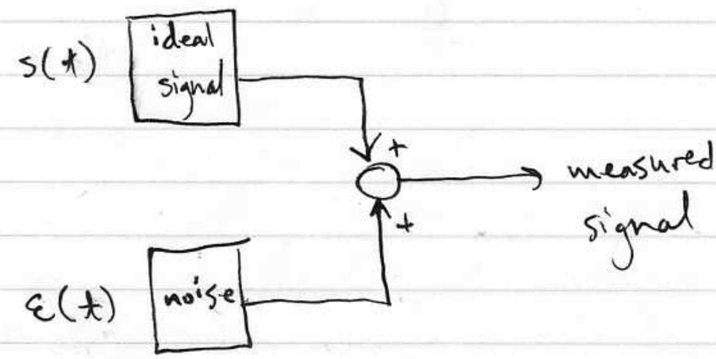
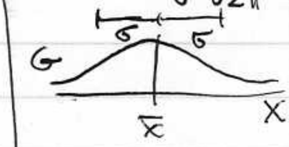
(1)

high freq $\rightarrow$ "AC" components from electrical theory (Alternating current) (5)



Gaussian:

$$G(x) = \frac{1}{\sigma\sqrt{2\pi}} e^{\left(-\frac{(x-\bar{x})^2}{2\sigma^2}\right)}$$



just

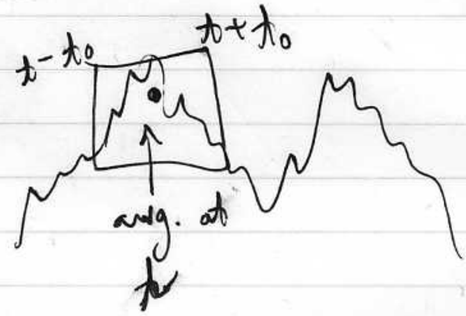
if we assume $\varepsilon(t)$ – Gaussian

- average $\bar{\varepsilon} = 0$

- that's why avging mult. pts. eliminates noise

$$\boxed{\int_{-\infty}^{\infty} \varepsilon(t)\, dt = 0}$$

Ex



last time we saw the discrete form

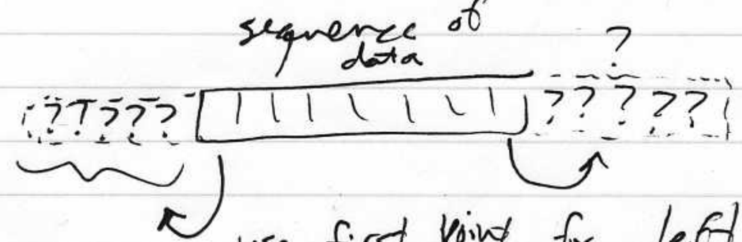
$$\boxed{x_f(i) = \frac{1}{2k+1} \sum_{j=i-k}^{i+k} x(j)}$$

example

Continuous:

$$\boxed{x_f(t) = \int_{t-t_0}^{t+t_0} x(\tau)\, d\tau}$$

what about the ends? No data



use first point for left edge,
last point for right edge (repeat)

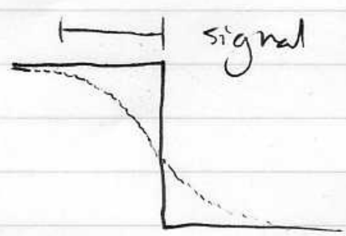
$$x(-j) = x(0) \quad \forall j < 0$$

→ creating the function in matlab
- example of sound → itunes and equalizer for band pass filters

→ if true make sound, play now

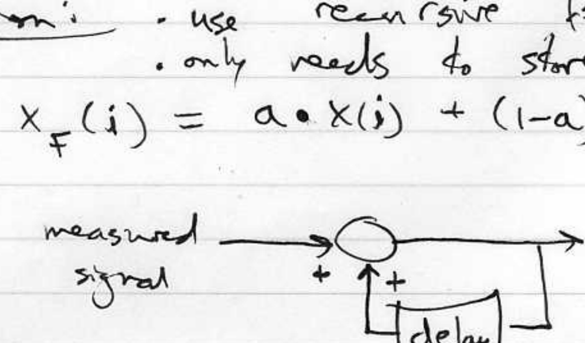
disadvantage(1) → all data must be in memory

disadvantage(2) → when we want to know when an event happens precisely, this filter can distort that.



Solution: . use recursive filters
. only needs to store one value from history.

$$x_F(i) = a \cdot x(i) + (1-a) \cdot x_F(i-1)$$



delay - delays output by one sample

3

5

easy to implement

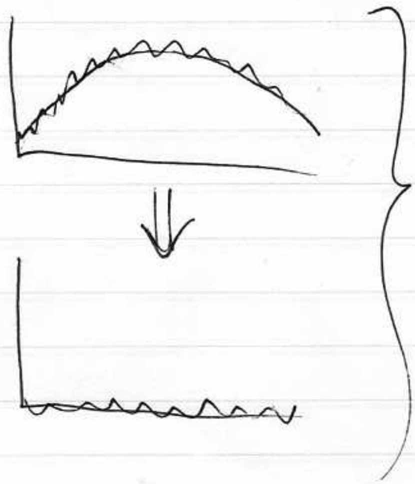
"a" determines 'strength' of filter

$(0 \le a < 1)$ → range of "a", stronger near 0

what if we want to get rid of low frequencies (DC) → ("Direct current")

high-pass

- we want to get rid of low frequencies



high pass filter

Find average (low frequency components) and subtract from original signal.

$$\boxed{X_{F_H} = X - X_{F_L}}$$

→ use any low pass filter to find L.F. components, use a large window size

- example