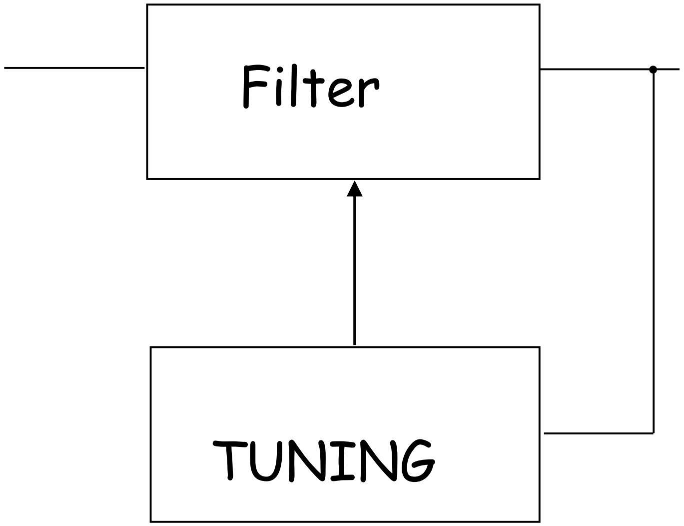
TUNING

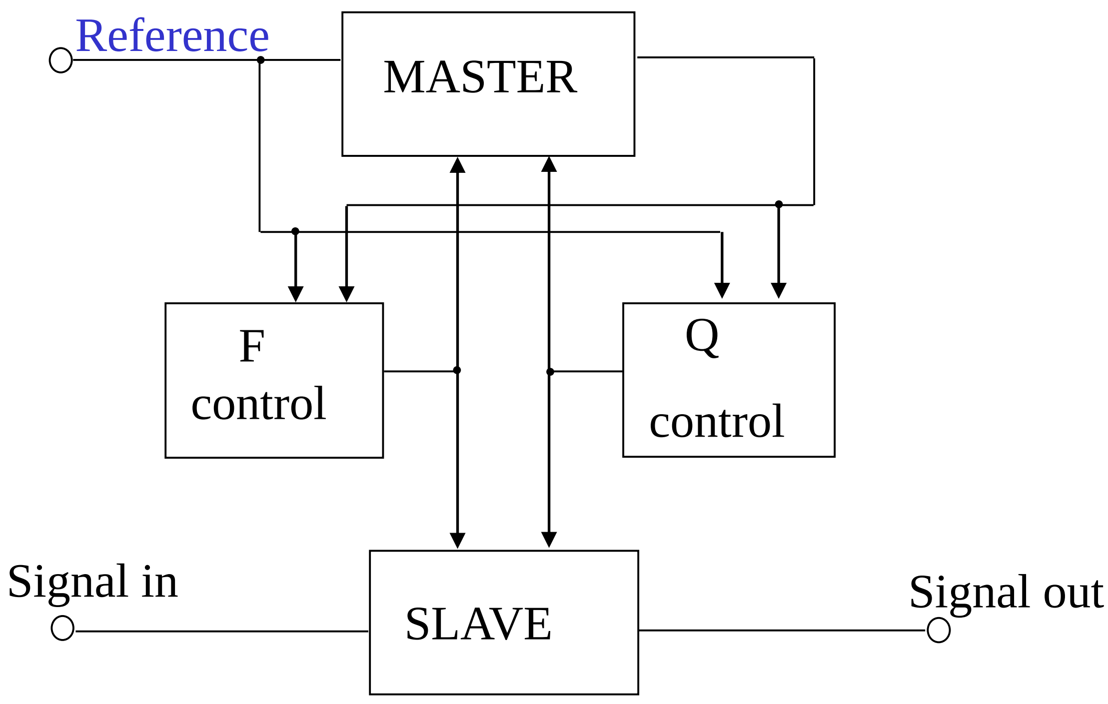
- Frequency characteristics of continuous-time filters are based on RC or LC products, or on G_m/C ratios, depending on the implementation
- Very accurate element values must be realized and maintained during filter operation in order to assure filter performance
- Due to fabrication tolerances, parasitics and environmental changes those accurate components are not normally available
- The generally adopted solution is to design an on-chip automatic tuning circuitry, which should evaluate the filter characteristics, compare them with references and apply correction signals to the filter
- However, extra costs of noise, power consumption and area should be minimized.



- To tune a filter's characteristics, (center frequency and quality factor) a known reference is needed. A review of the literature shows that an accurate reference frequency (a system clock) has been agreed upon among designers as the most reliable standard for frequency tuning

- The main purpose of the filter is to process the information carrying signal. Since applying both references and each signals to the filter will cause undesirable interference, a control system should be constructed for proper filter operation

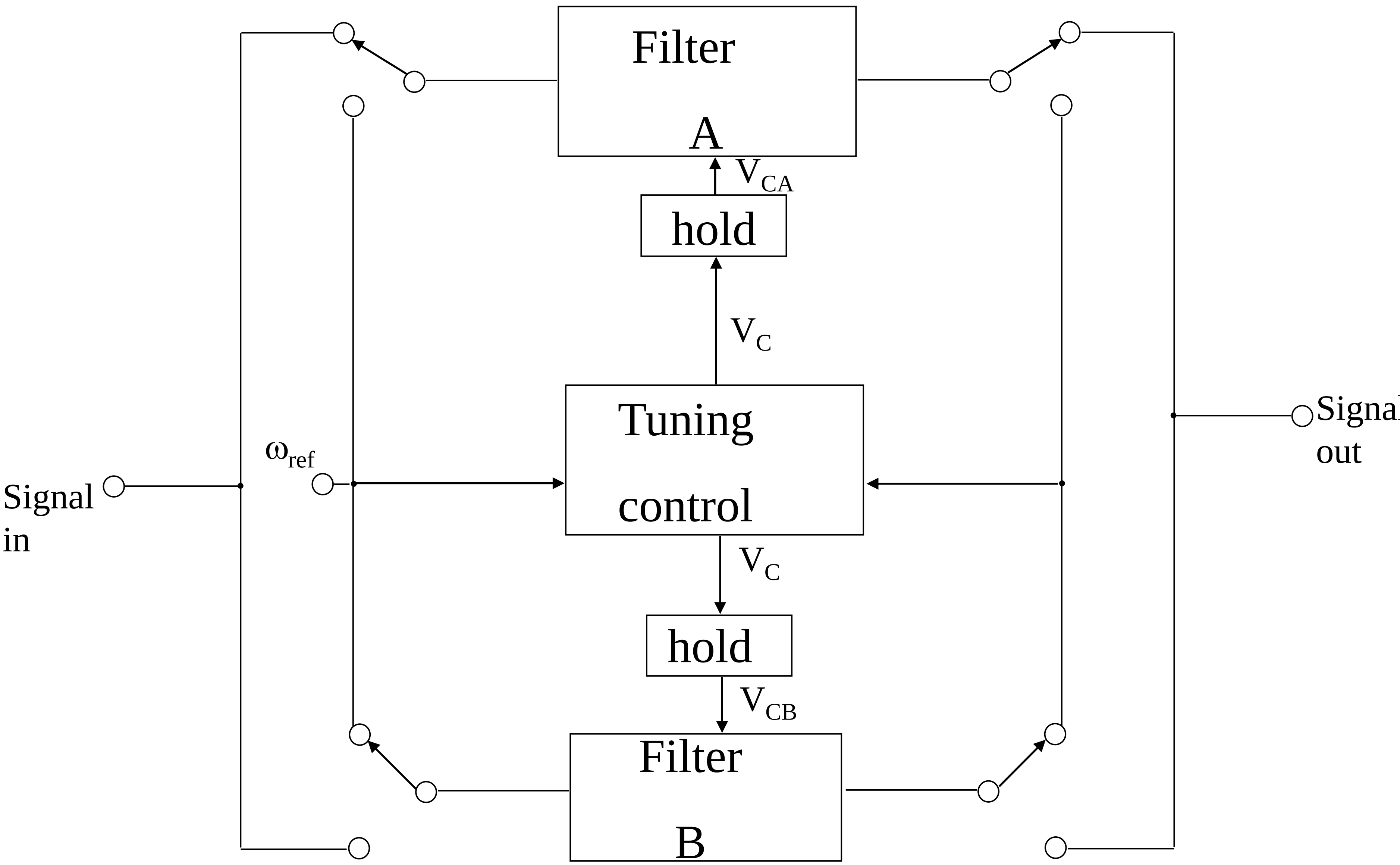
* <u>MASTER-SLAVE tuning</u>



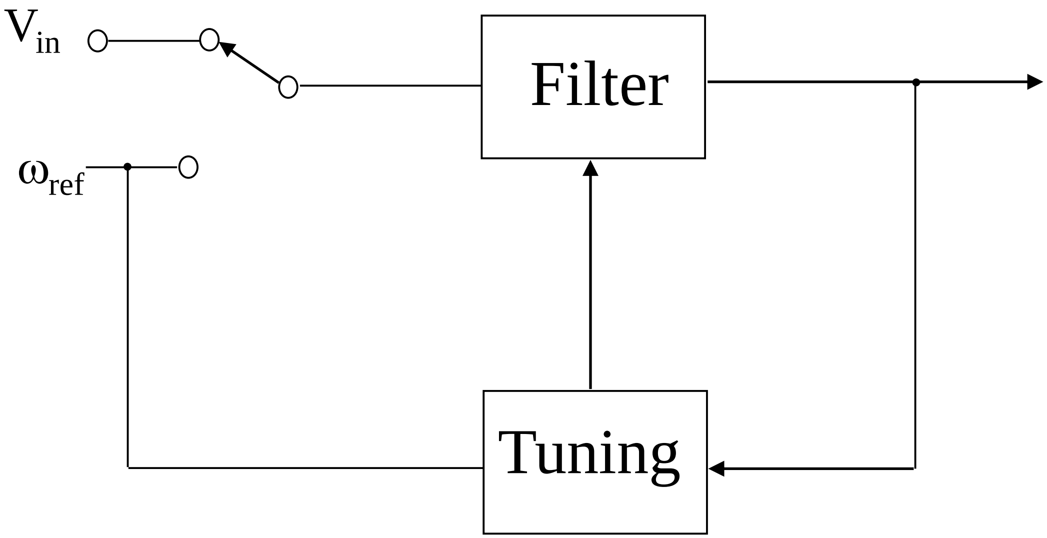
- In this approach the reference signal is applied to a so-called master filter, which models all relevant performance criteria of the main (or slave) filter.
- Master can be an integrator, or the whole biquadratic filter.
- Since the response of slave is not evaluated, (only the master is tuned) matching between master and slave is critical for tuning accuracy.
- For better matching, master and slave filters should be physically near each other.
- However, being close increases feedthrough of control signals to the main filter, so the physical distance between master and slave should be a good compromise between good matching and low noise interference.
- Furthermore, if an integrator is used for the master, G_m/C ratios (or capacitor ratios) determine the filter characteristics. Parasitic effects should be minimized.

OFFLINE TUNING

- Since matching is the bottleneck for master-slave, it may not be suitable for all applications.
- Offline tuning is based on time sharing, in which tuning is performed when the filter is not used to process the main signal. After tuning, appropriate control voltages are kept fixed until the next period.


Filter
A
V_{CA}
hold
V_C
Tuning
control
ω_{ref}
Signal
in
Signal
out
V_C
hold
V_{CB}
Filter
B

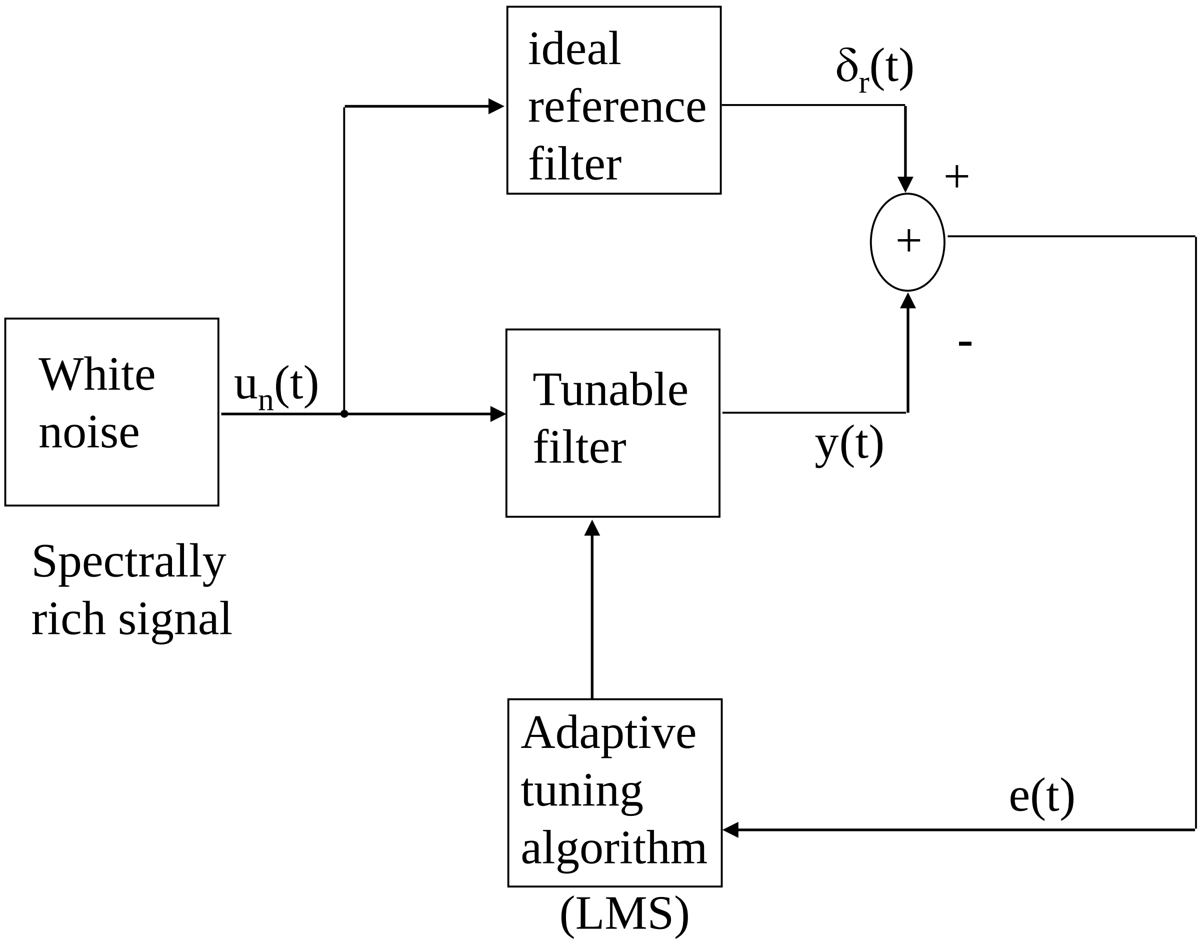
- Switching is a problem.
- At high frequencies, switching noise, feedthrough increases
- Even at low freqencies, switching may degrade the signal.
- Filter is actually tuned.
- Depending on the application, switching may not be required.



- However, proper operation cannot be guaranteed since the filter is not continuously tuned.

ADAPTIVE FILTER TECHNIQUE

- This technique is based on model-matching system shown below:



- Applying a spectrally rich signal $u_n(t)$ to the filter, an adaptive algorithm (LMS) is used to adjust the coefficients of the tunable-filter so as to minimize the error signal e(t) where

$$e(t)=\delta_r(t) - y(t)$$

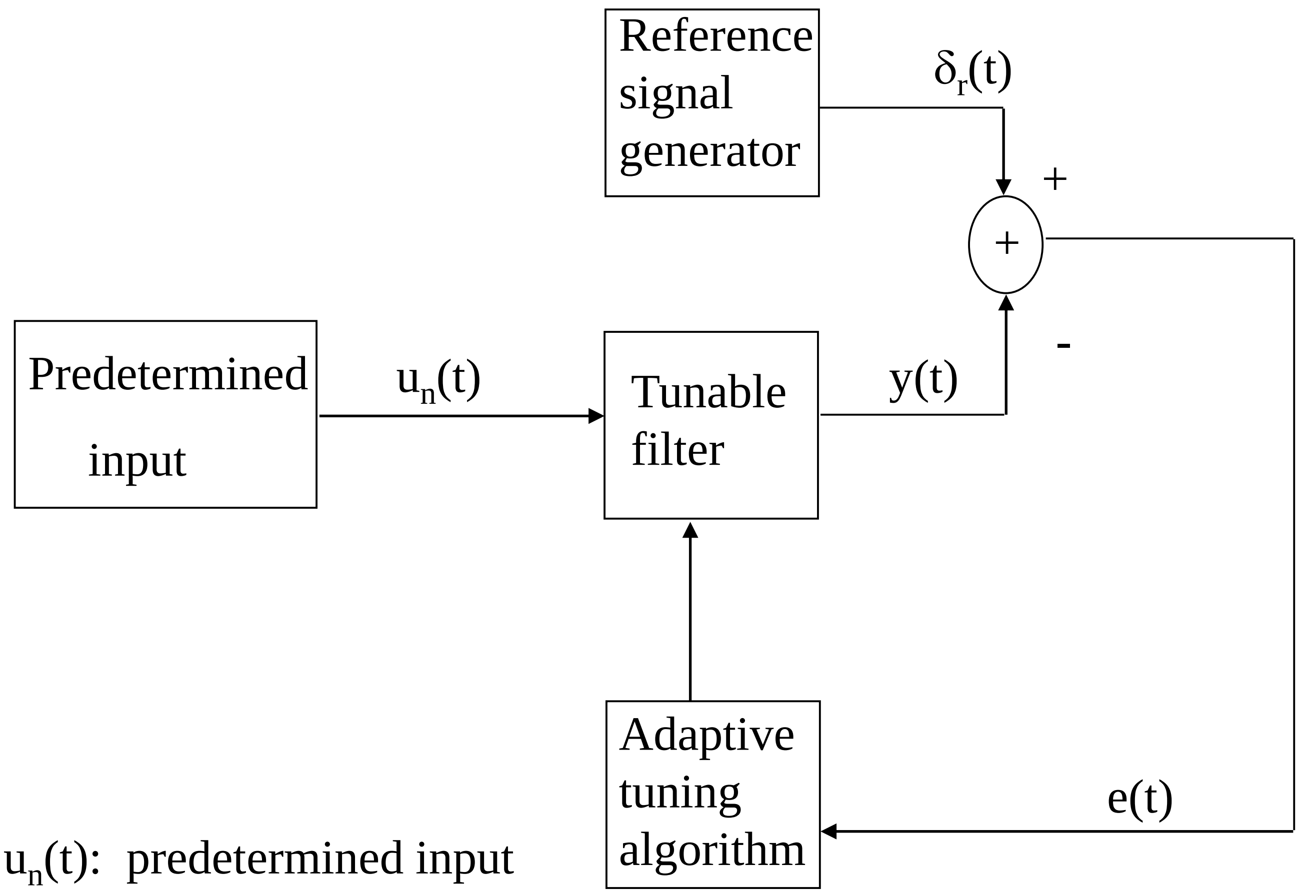
Output of ideal reference filter

Tunable filter output

- Since the filter continuously services output, white noise source is replaced by the signal to be processed in the actual implementation.

- However, it is assumed that the spectral content of the input signal is sufficiently rich to fully characterize the system.

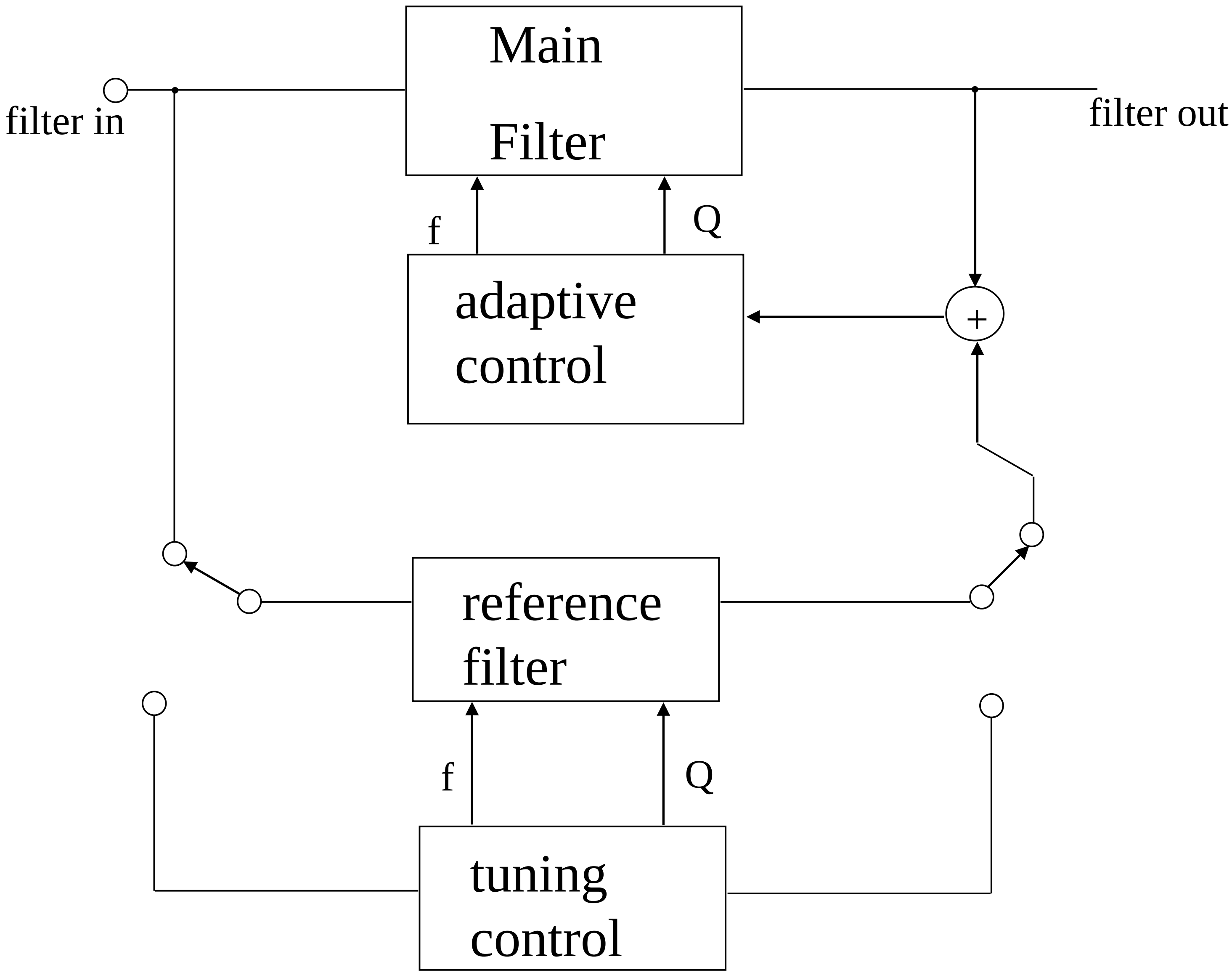
- The assumption of "ideal reference filter" obviously conflicts the fundamental need for tuning. To achieve the ideal filter, an adaptive tuning system is utilized.



$u_n(t)$: predetermined input

$\delta_n(t)$: pre-calculated output

- The overall picture



- Therefore, the main filter is tuned to the same level of accuracy achieved by self-tuning of the reference filter while continuously processing the desired signal
- Complex, power and area consuming.

OTHER TECHNIQUES

- The reference can be applied as a common-mode signal to a pseudo-differential circuit
- At the output, the main signal, which is applied differentially, can be recovered
- No tracking problem, but interference may be trouble
- Orthogonal reference tuning: Reference and processed signal can be separated at the filter output. However, characteristics of the input signal has to be known (for orthogonality).

Tuning techniques (Summary)

* Master-slave
* Offline
* Adaptive
* Orthogonal reference and common-mode reference

Frequency tuning:

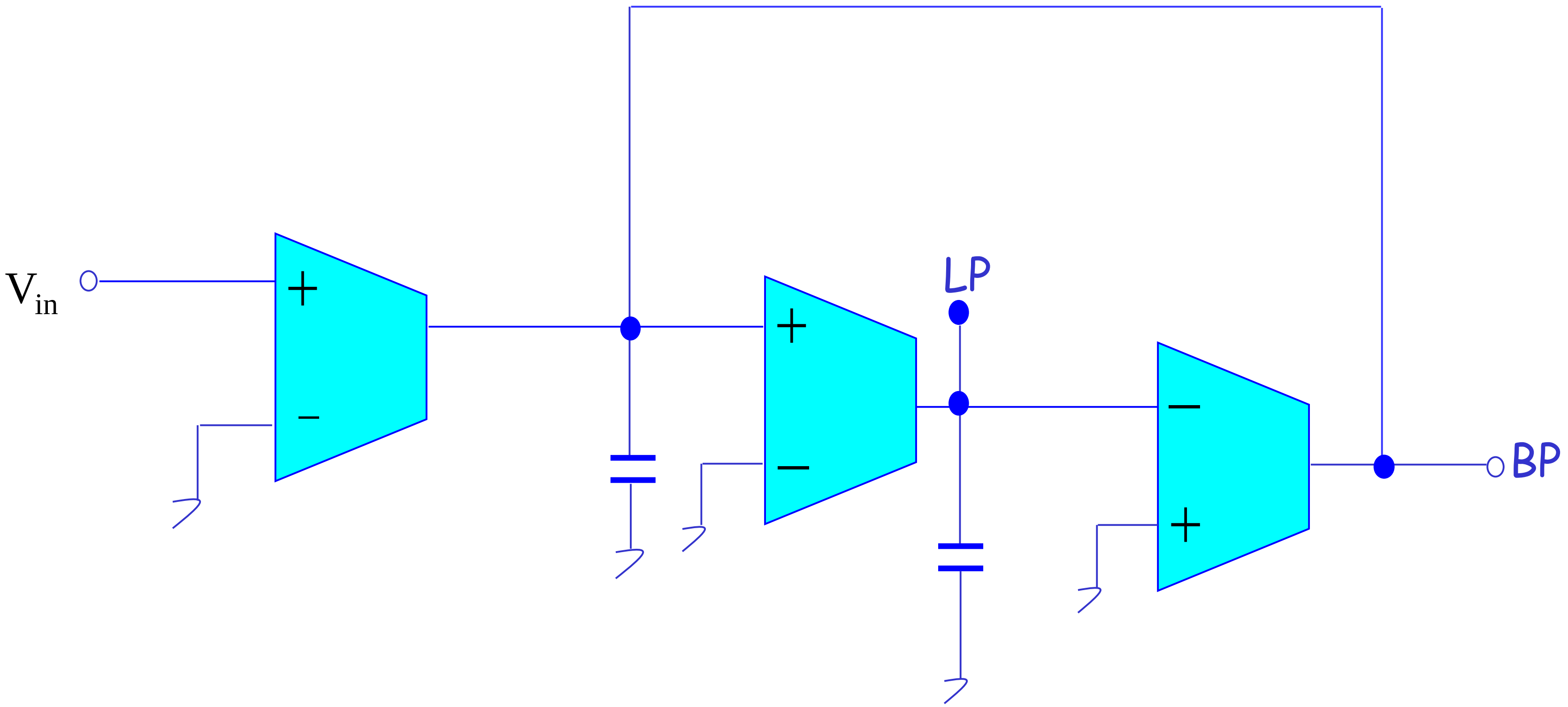
General biquadratic filter

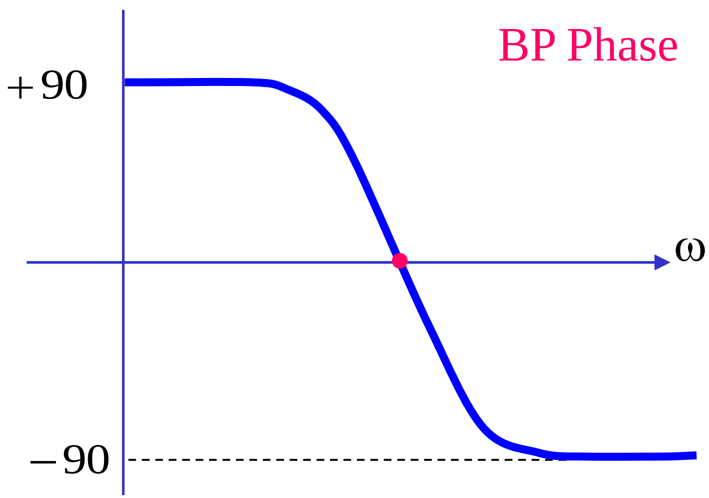
$$\frac{N(s)}{s^2 + \frac{\omega_o}{Q}s + \omega_o^2}$$

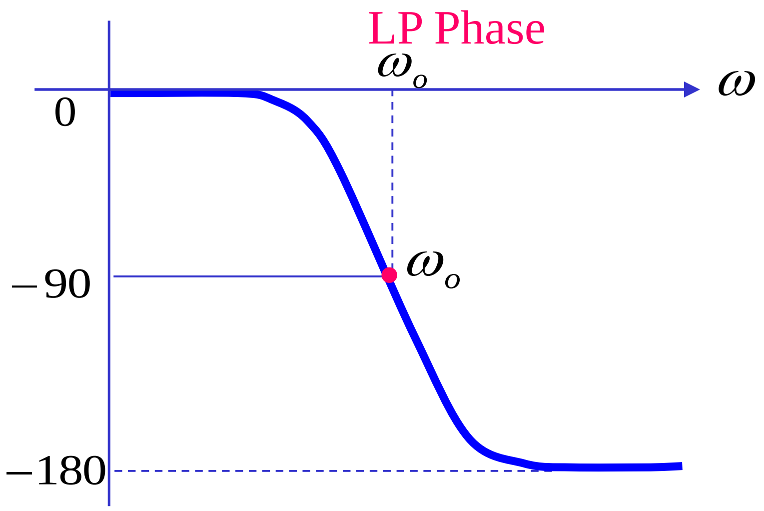
$N(s)=\omega_o^2 \Rightarrow$ lowpass

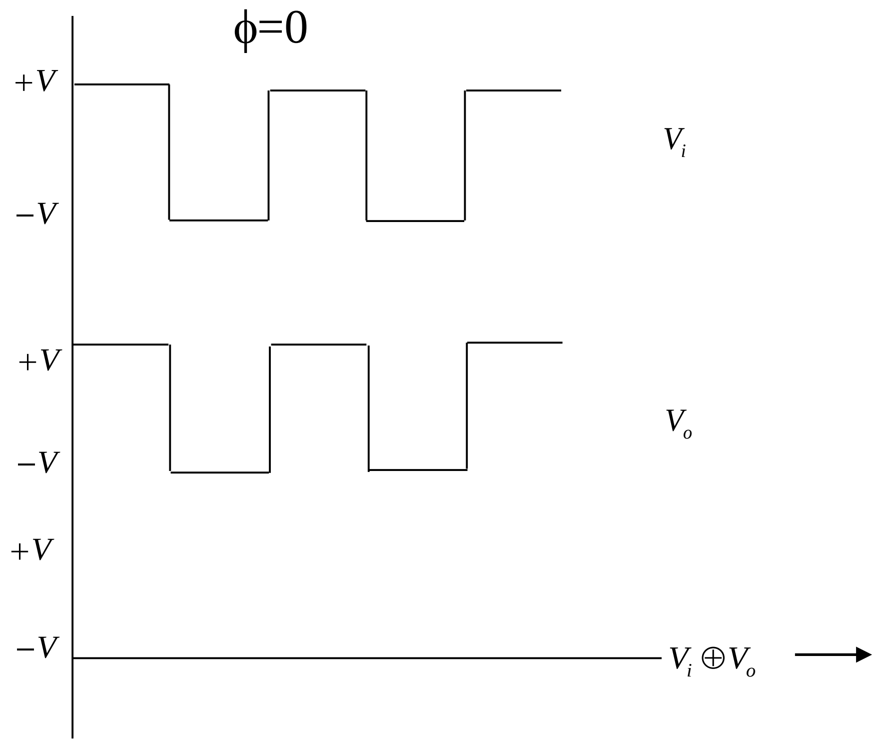
$N(s)= \omega_o \cdot s \Rightarrow$ bandpass

$N(s)= s^2 \Rightarrow$ highpass

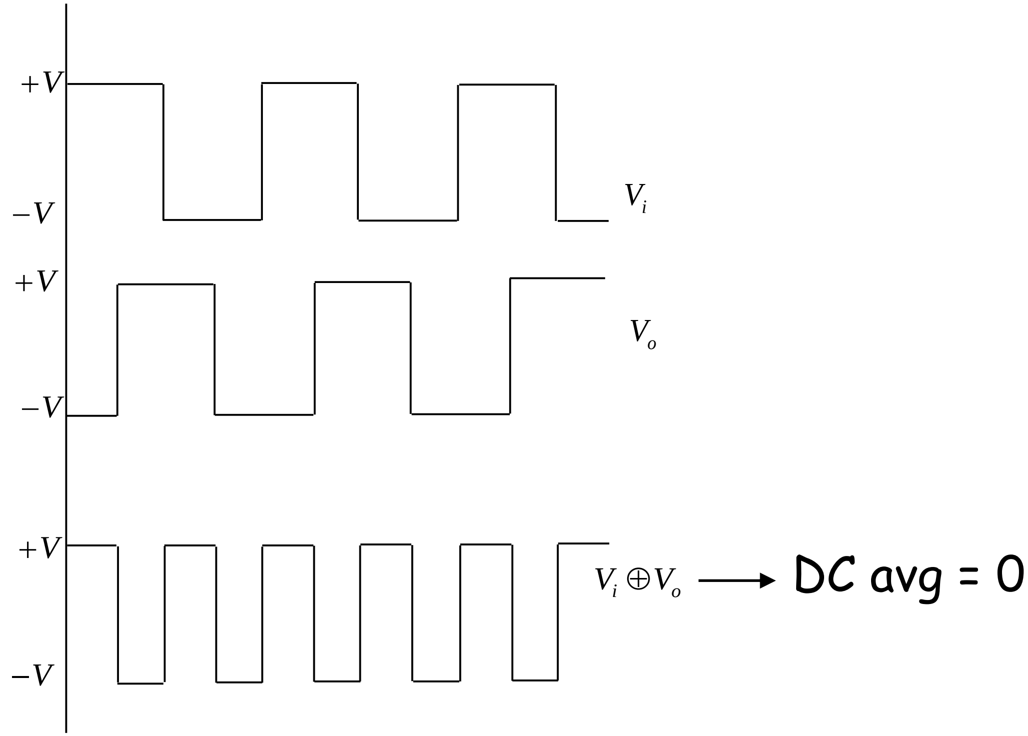


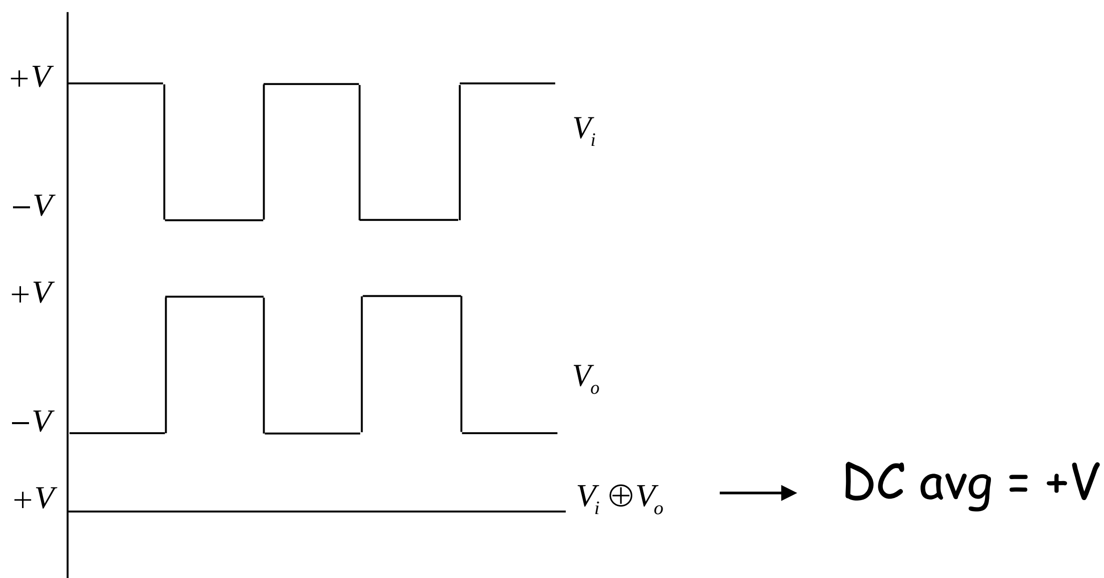




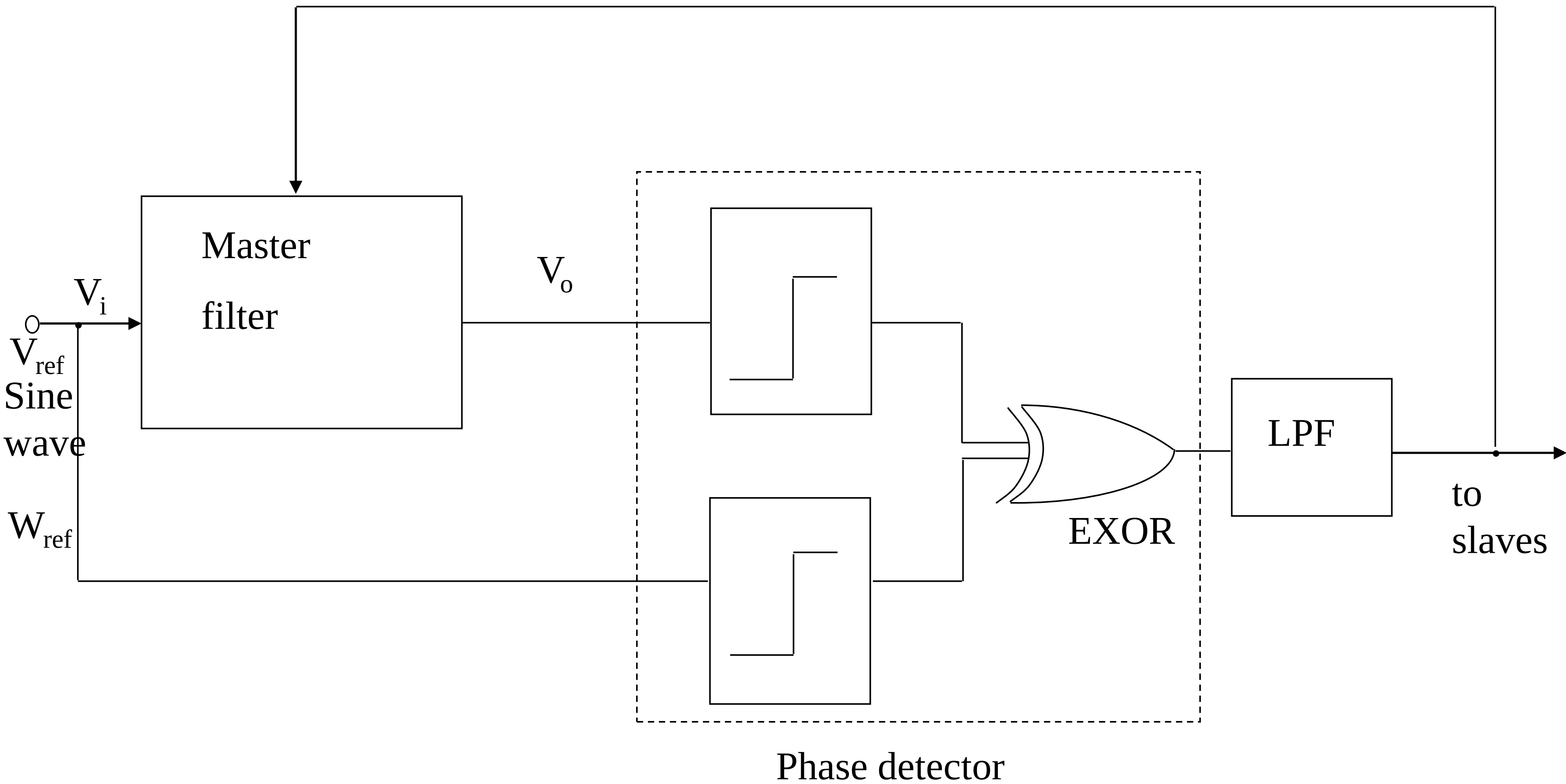


DC avg = -V

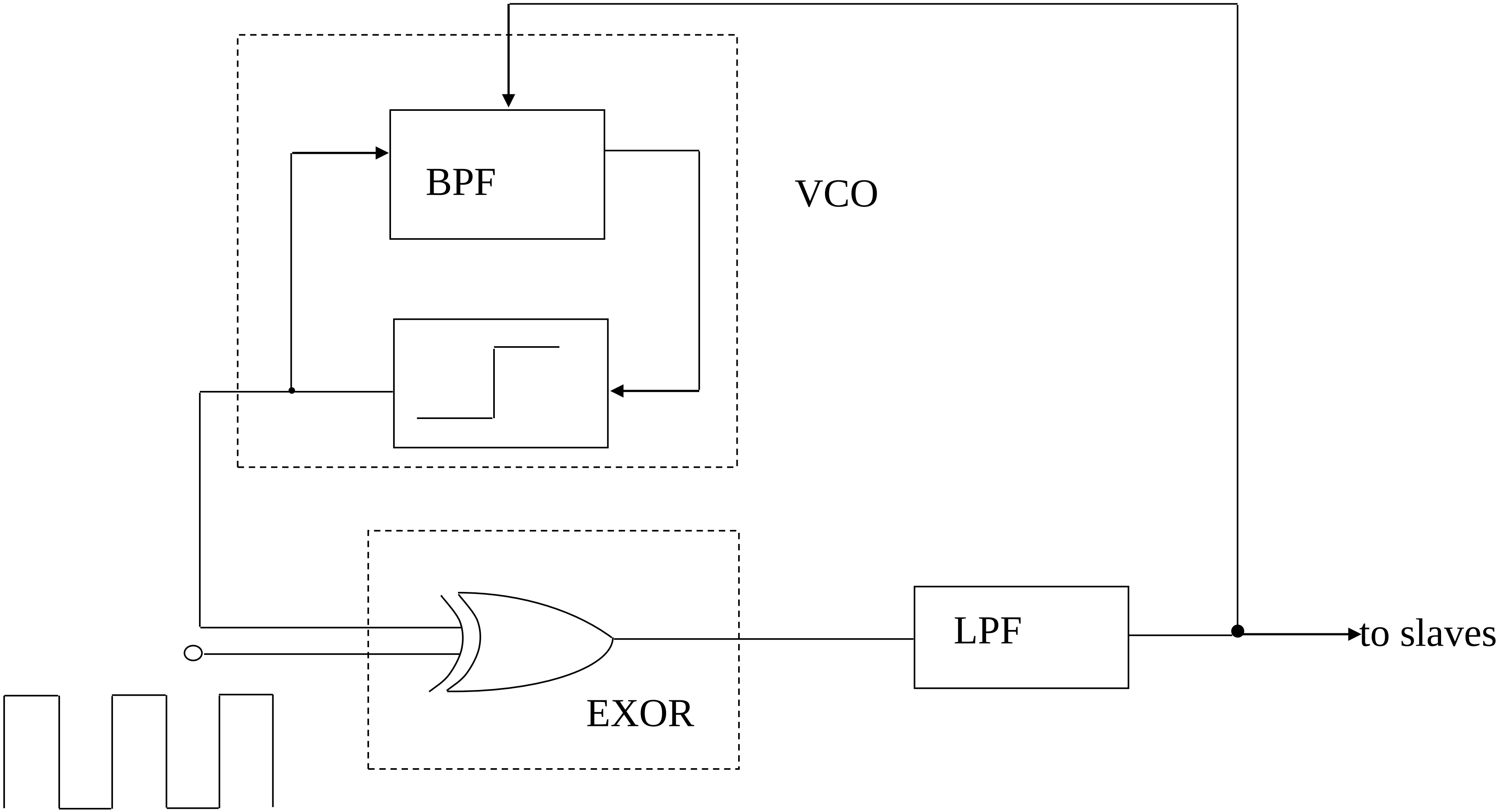




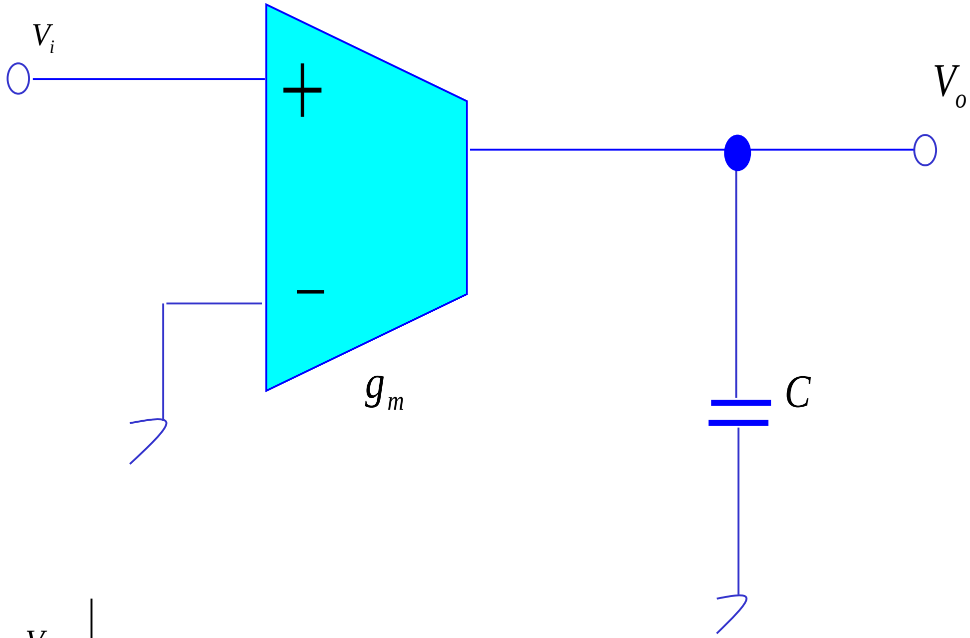
If we form a feedback loop using $V_i \oplus V_o$, we can tune the center frequency to ω_o



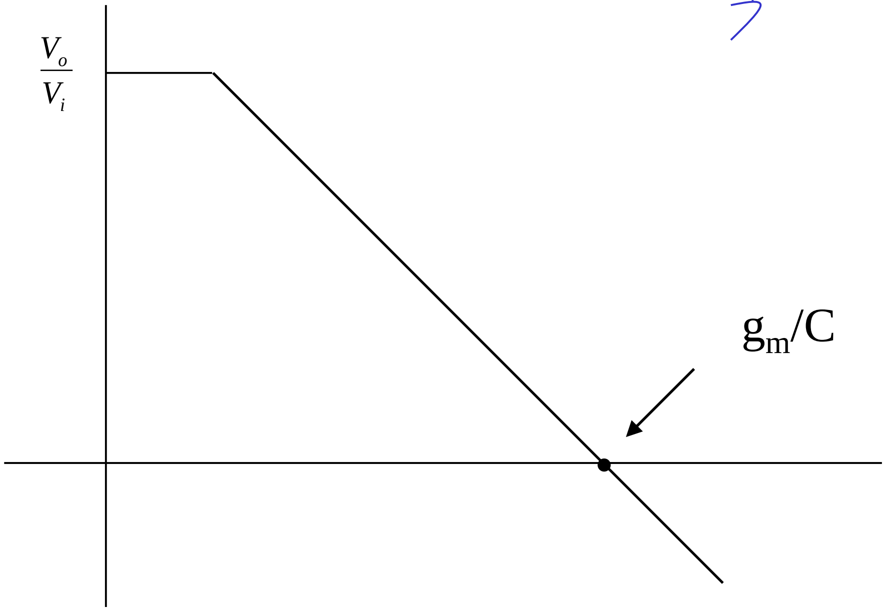
Frequency-locked loop



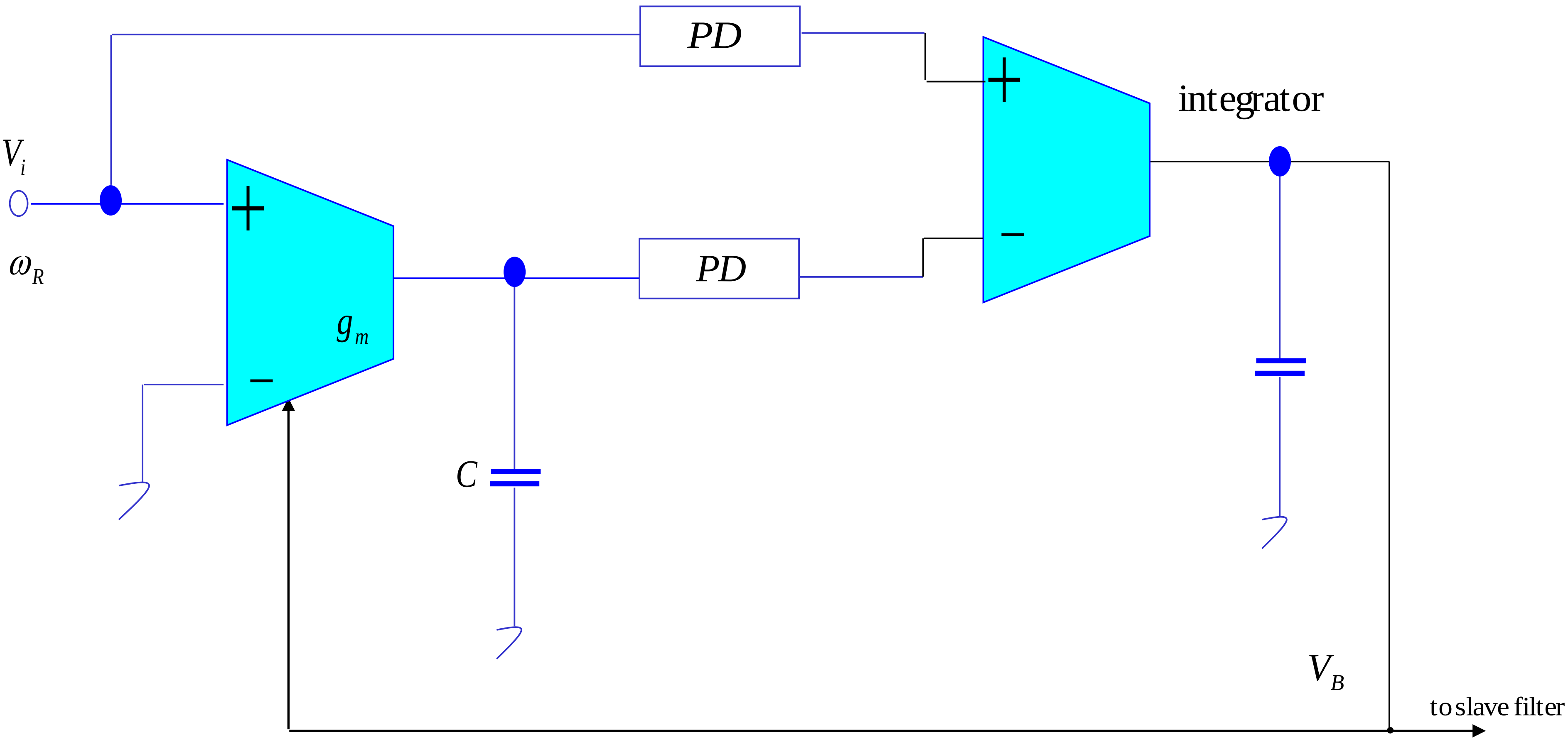
Phase locked loop



V_i
+
−
g_m
V_o
C

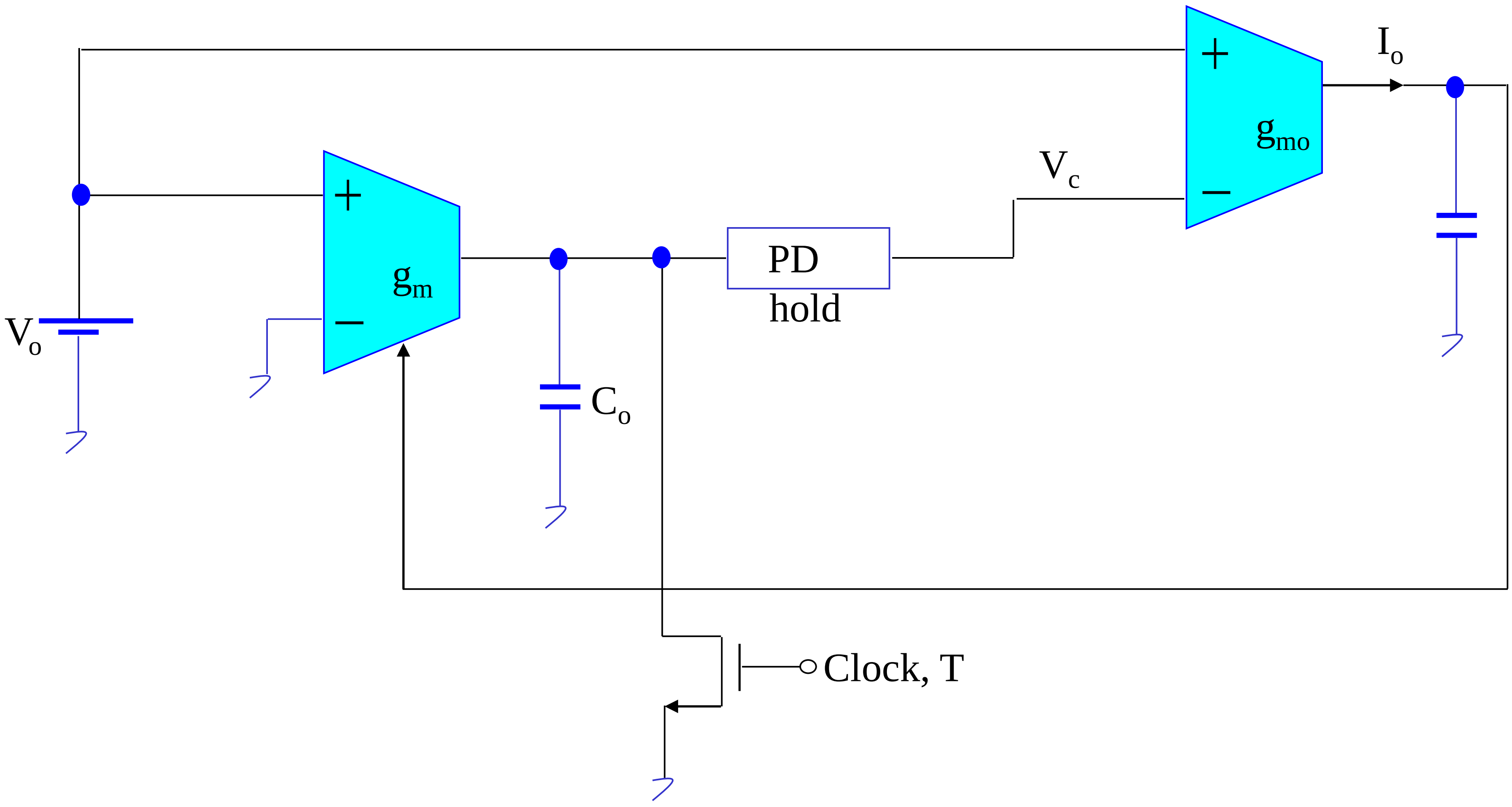


$\frac{V_o}{V_i}$
g_m/C

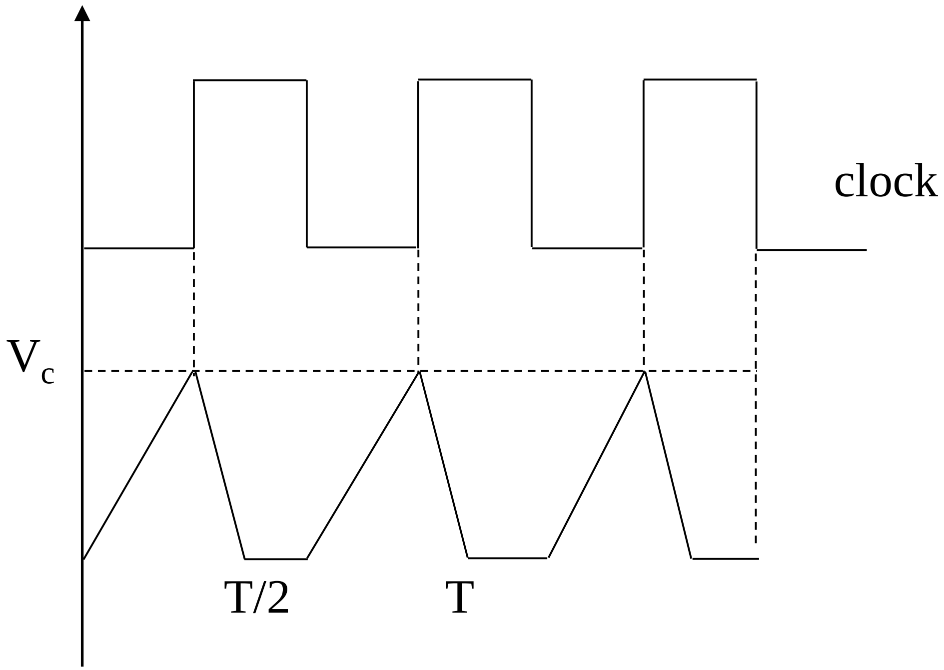


$$g_m / C \rightarrow \omega_R$$

PD : Peak Detector



+
−
g_{mo}
I_o
V_c
+
−
g_m
PD
hold
V_o
C_o
Clock, T



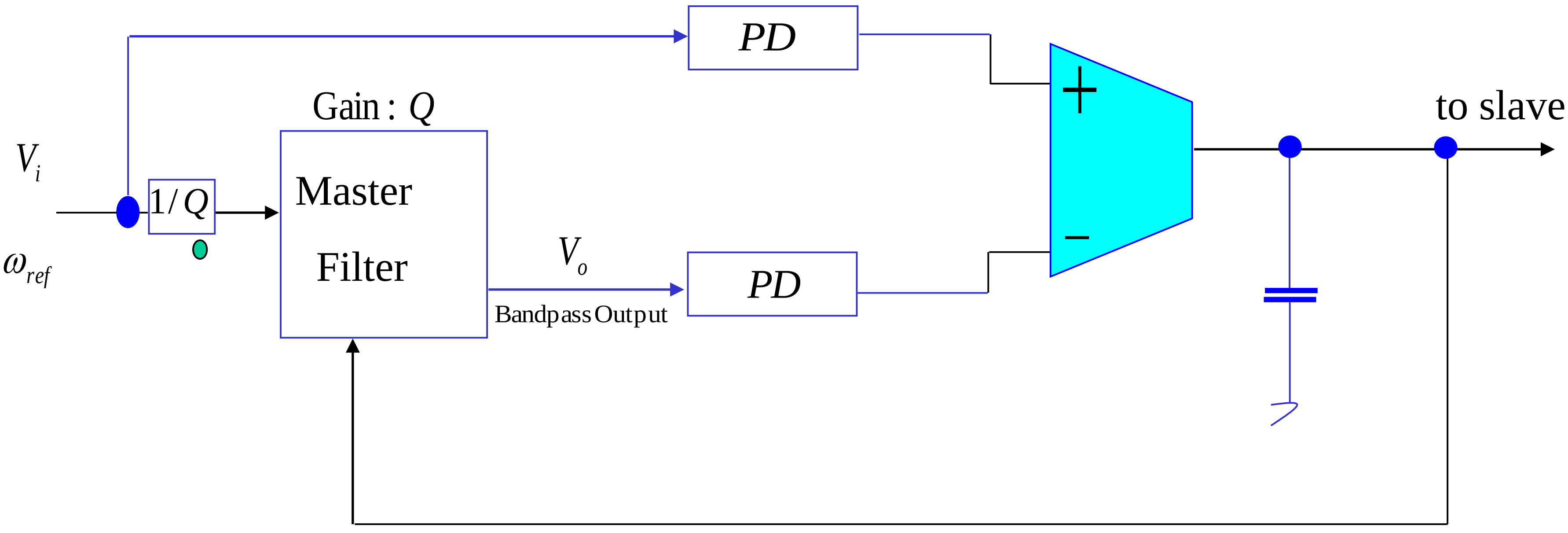
$$I_o = g_{mo}(V_o - V_C) = g_{mo}(V_o - \frac{T}{2}\frac{g_m}{C_o}V_o)$$

Loop will converge when I_o=0

$$\gamma \frac{T}{2}\frac{g_m}{C_o} = 1$$

$$\boxed{\frac{g_m}{C_o} = \frac{2}{T}}$$

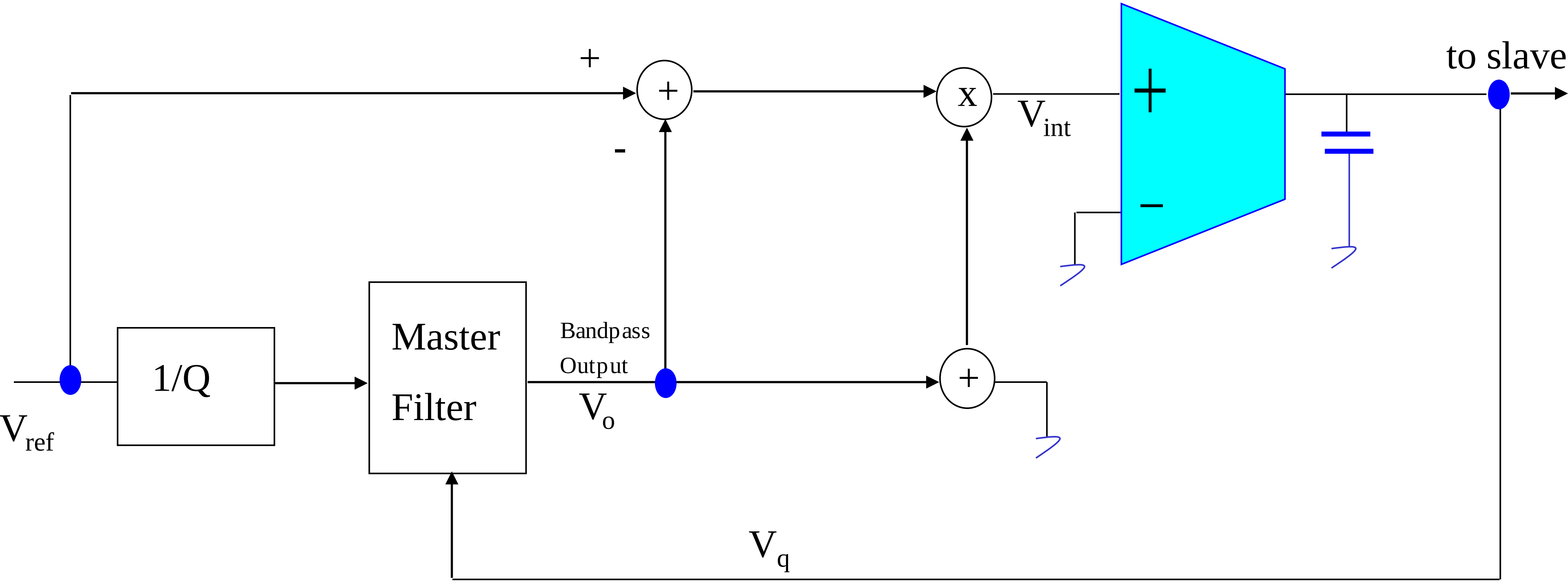
Q-tuning



Magnitude locked-loop

V_i and V_o magnitude are equal in steady state

filter passband gain=Q



$$V_{int} = V_o\,(V_i - V_o)$$

$$V_o = V_i \Rightarrow V_{int} = 0 \quad \text{tuned}$$

$$\dot{V}_q(t) = \mu (V_{in} - V_{bp}) V_{bp}$$

LMS Algorithm Derivation.- The mean square error (MSE) is defined as $E(t)=0.5[e(t)]^2 = 0.5[d(t)-y(t)]^2$

where d(t) is the desired output signal, and y(t) is the actual output signal. The steepest descent algorithm is defined as:

$$\frac{dW}{dt} = -\mu \frac{\partial E}{\partial W}$$

$$\frac{dW}{dt} = -\mu \frac{\partial E}{\partial y} \frac{\partial y}{\partial W}$$

$$\frac{dW}{dt} = -\mu \frac{\partial [0.5\{d(t) - y(t)\}^2]}{\partial y} \frac{\partial y}{\partial W}$$

$$\frac{dW}{dt} = \mu [d(t) - y(t)] \frac{\partial y(t)}{\partial W}$$

$$\dot{W} = \mu [d(t) - y(t)] G(t) = \mu e(t) G(t)$$

Linear System case.

$$y(t) = \sum_{i=0}^{n} w_i x_i,$$

where:

x_i is the input signal.

Therefore:

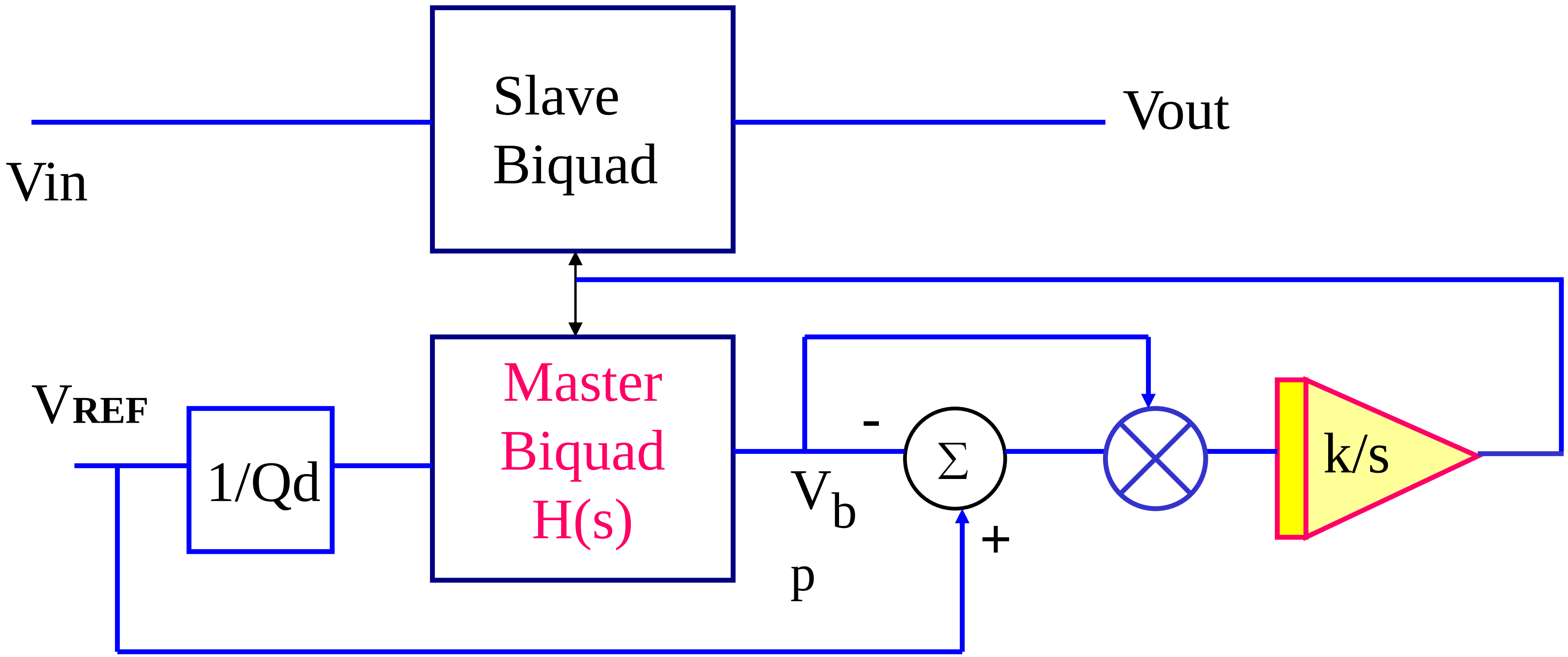
$$\frac{dW}{dt} = \mu[d(t) - y(t)] \frac{\partial \sum_{i=0}^{n} w_i x_i,}{\partial W} = \mu[d(t) - y(t)] x_i$$

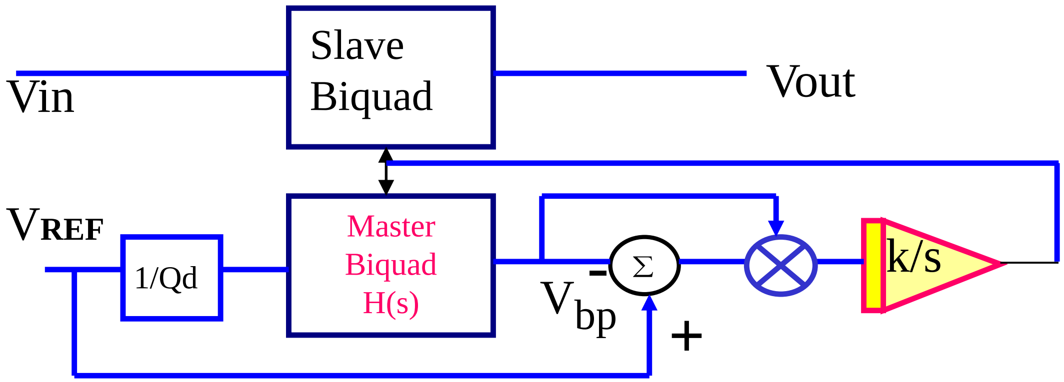
$$\dot{W} = \mu e(t) x_{xi}$$

Adaptive LMS Algorithm

$$\dot{w}_i = \mu\left[d(t) - y(t)\right]g_i(t)$$

Where is the tuning signal, d(t) is the desired response, y(t) is the actual response, and g_i (t) is the gradient signal (that is the direction of tuning.





Block Diagram Solution