
USC Viterbi
School of Engineering
*Center for Cyber-Physical Systems and the Internet of Things*

# SOFTWARE DEFINED RADIO

## USR SDR WORKSHOP, SEPTEMBER 2017

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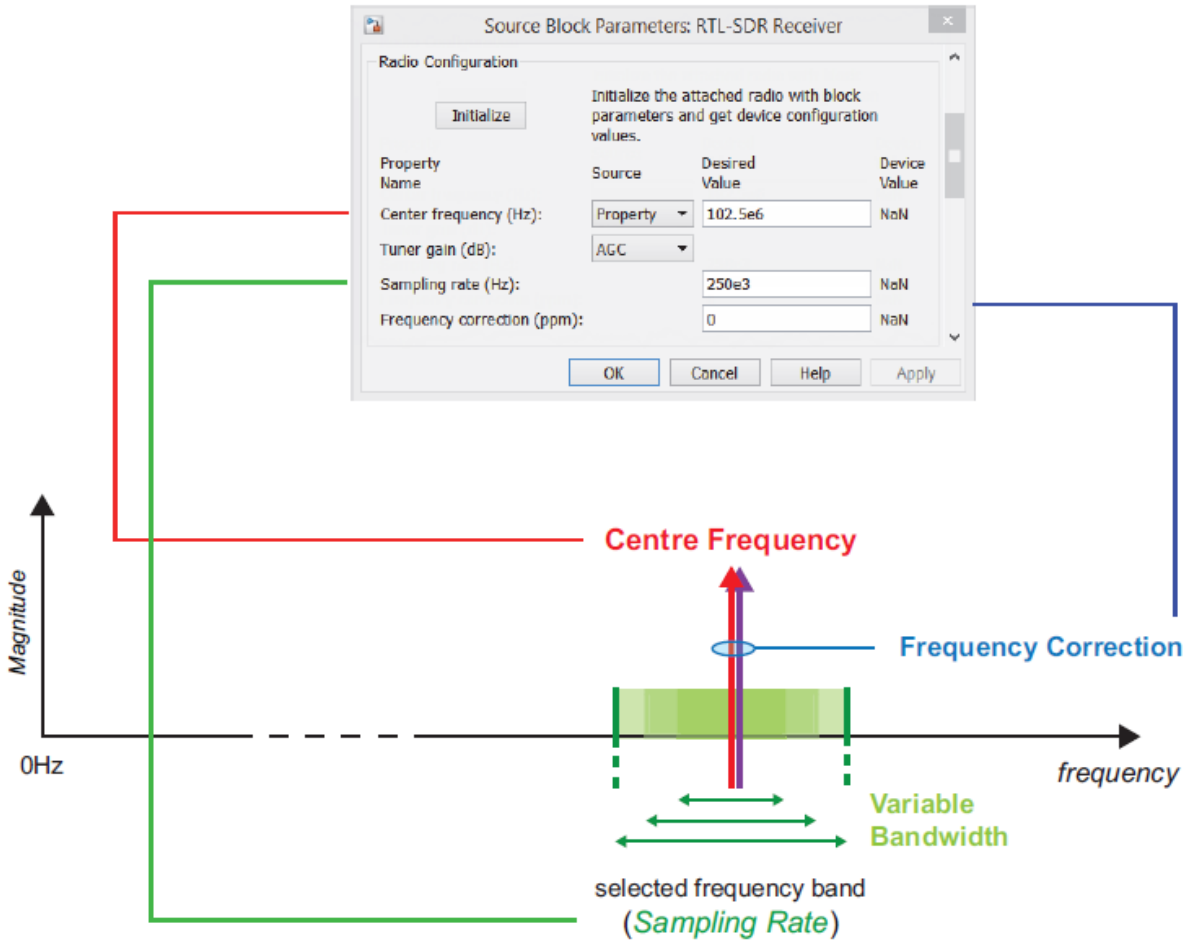
SESSION 3: PHASE AND FREQUENCY SYNCHRONIZATION

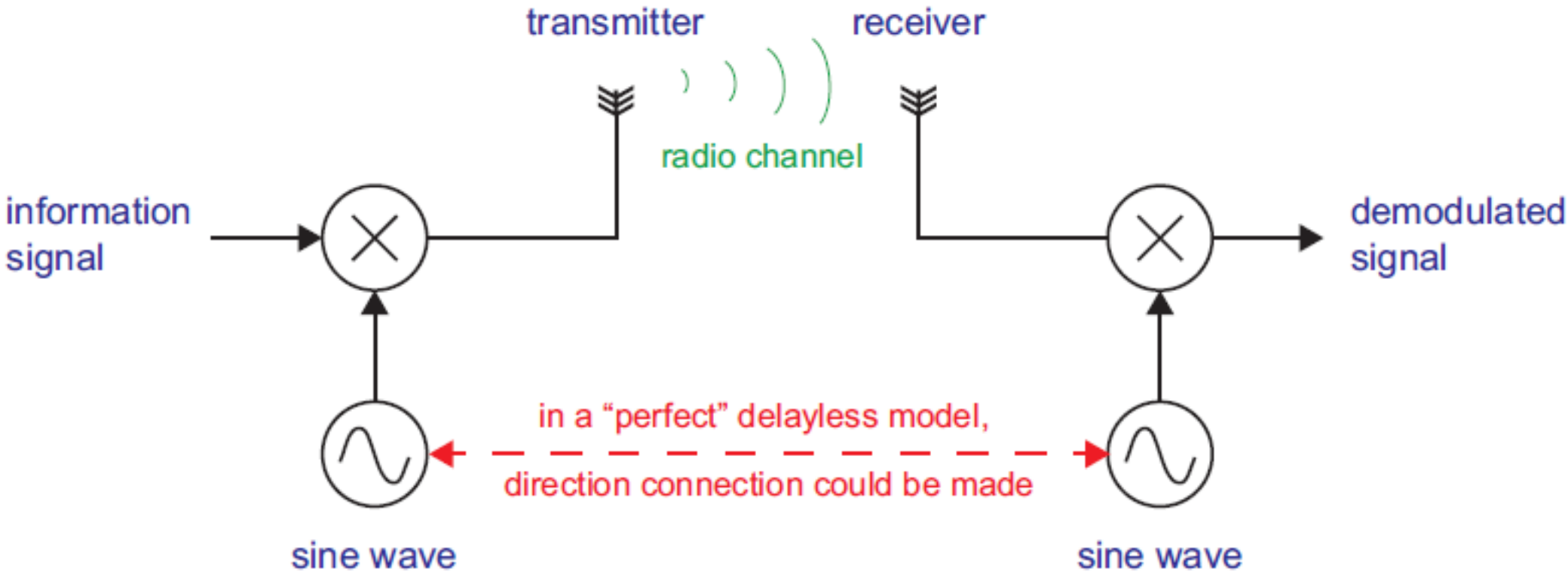
# TUNNING

- Tuning, consist on selecting the right value for the LO and the appropriated sampling rate.
- All the tuning parameters are setup on Simulink block.
- You should be carful with the LO difference between SDR boards.
- On B200, the sample rate is defined by the relation between clock and up/down sampling.



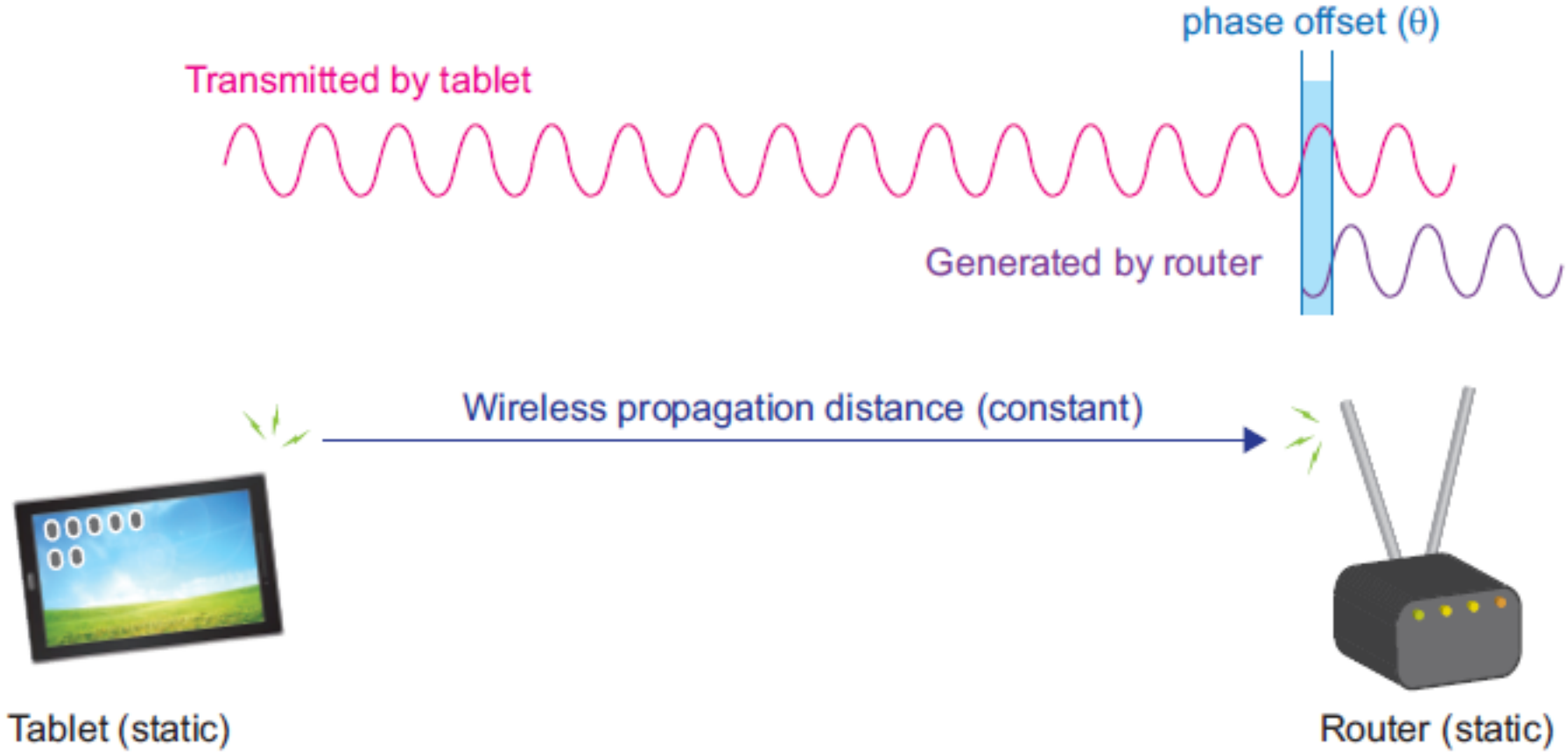
# THE SYNC PROBLEM

- Channel effects:  A) Propagation delay B) frequency shift
- If we could consider No delay, always we have difference between LOs.



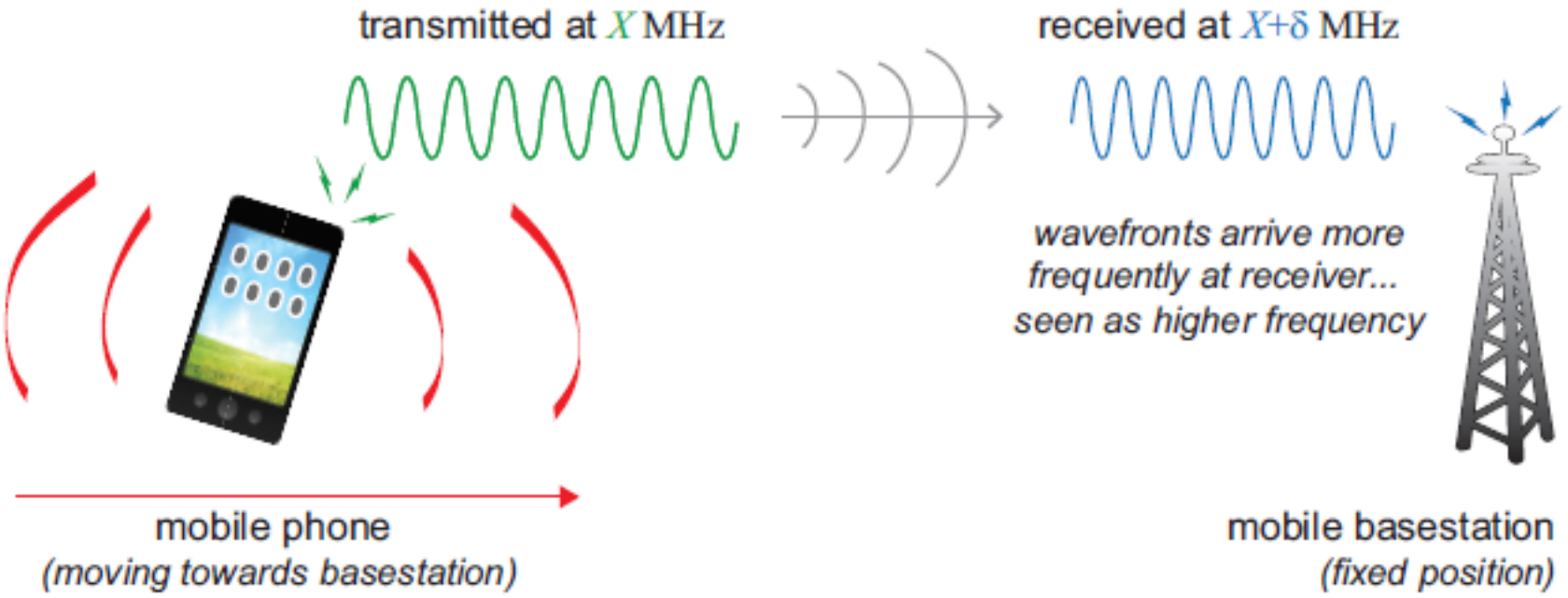
# PROPAGATION DELAY

- If TX ad RX are fixed position, we always have fixed phase offset.



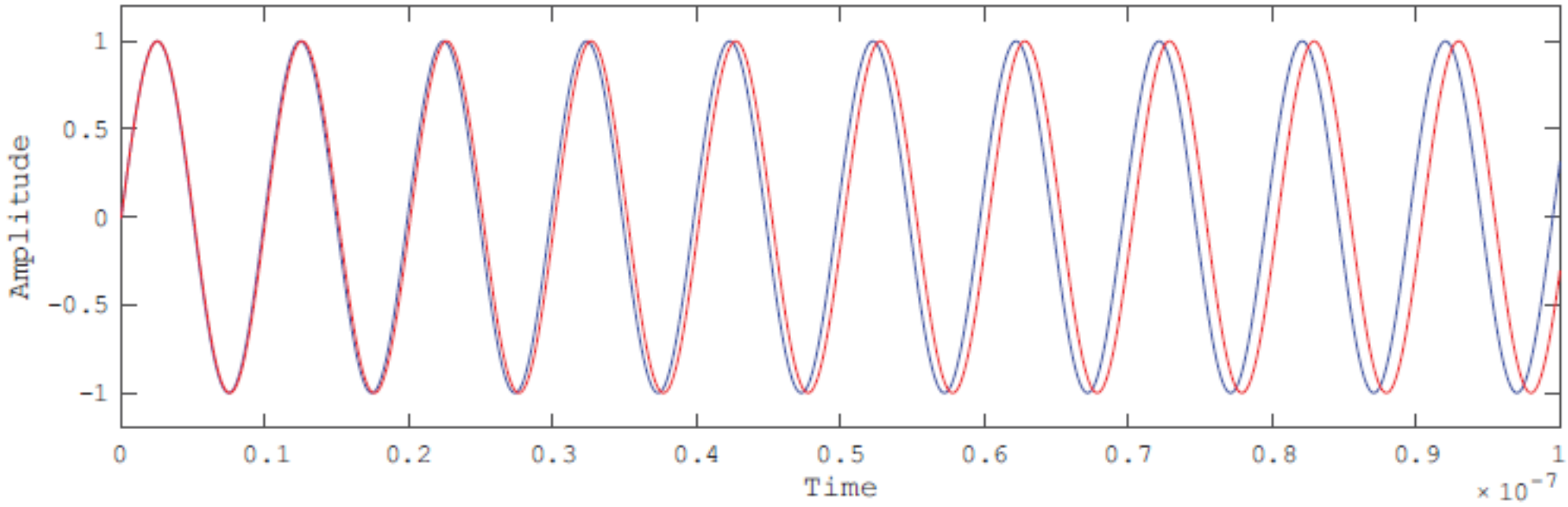
# FREQUENCY SHIFT

- If TX and RX moves, there will be a time variance phase shift that is equivalent to a frequency shift.
- Doppler effect.



# HARDWARE EFFECTS

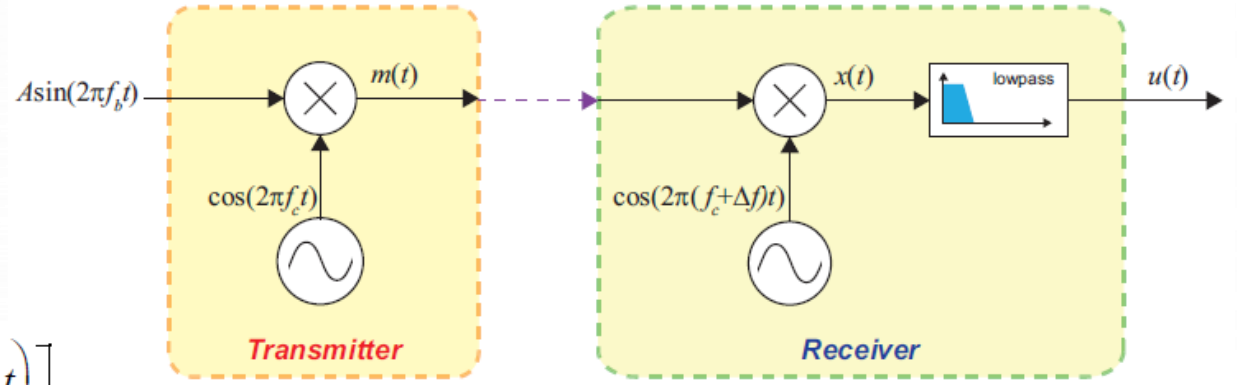
- LO change over time due to: temperature, manufacture process, ageing and others.
- Example: error of 5000ppm, f1=100,5Mhz, f2=99,5Mhz, Fc=100Mhz



# COHERENT DEMODULATOR

- Demodulate with an LO that is not exactly the same….problems, so **Solutions:**
- Option 1: TX the carrier like DSB-TC (pros/con)
- Option 2: **Recovering the carrier from the modulated signal.**
- Modulated Signal (DSB-SC)

$$m(t) = \frac{A}{2}\left[\sin(2\pi(f_c+f_b)t) - \sin(2\pi(f_c-f_b)t)\right]$$




- Demodulated Signal

$$x(t) = \cos(2\pi(f_c+f_\Delta)t) \times \frac{A}{2}\left[\sin(2\pi(f_c+f_b)t) - \sin\left(2\pi(f_c-f_b)t\right)\right]$$

lowpass filtered term

$$u(t) = \frac{A}{4}\sin\left(2\pi(2f_c+f_\Delta+f_b)t\right) - \frac{A}{4}\sin\left(2\pi(f_\Delta-f_b)t\right)$$

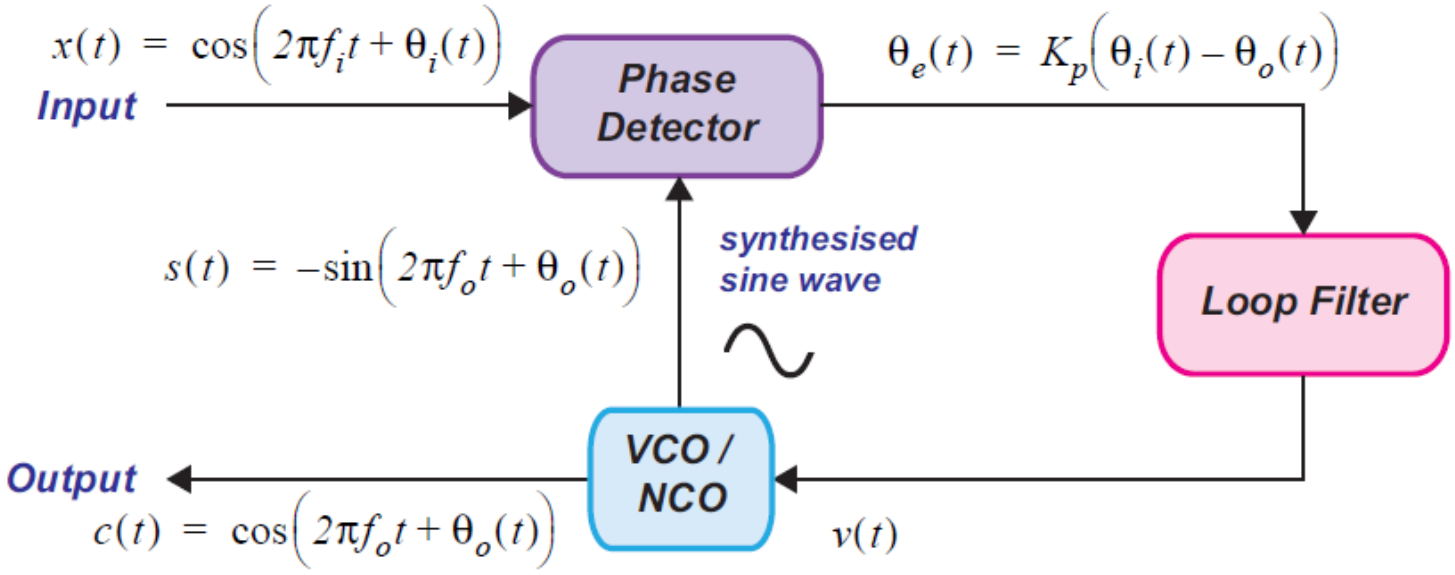
$$-\frac{A}{4}\sin\left(2\pi(2f_c+f_\Delta-f_b)t\right) + \frac{A}{4}\sin\left(2\pi(f_\Delta+f_b)t\right)$$

lowpass filtered term

$$u(t) = \frac{A}{4}\left[\sin(2\pi(f_b+f_\Delta)t) + \sin(2\pi(f_b-f_\Delta)t)\right]$$

# PHASE LOCK LOOP, REVIEW

- The PLL is the fundamental component on every coherent receiver.
- PLL:
  - Phase Detector: the output is proportional to the phase difference between received signal and locally generated.
  - Controlled Oscillator: it is a VCO for analog receiver and NCO for digital receiver.
  - Loop filter: Filter acts upon the output of the Phase Detector to remove unwanted high frequency terms, and produce the signal that drives the VCO or NCO.

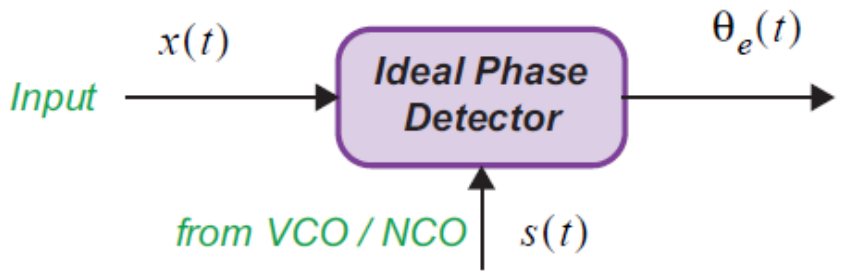


# PHASE DETECTOR

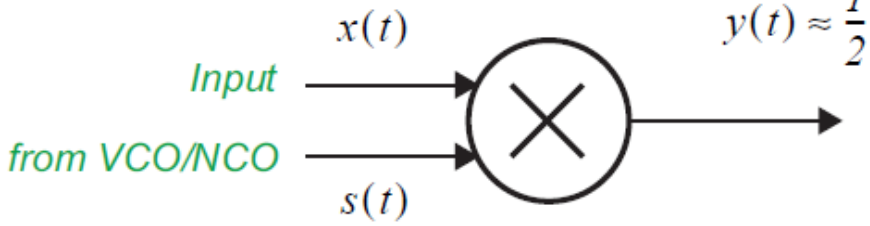
- The phase error is proportional to the phase difference:

$$\theta_e(t) = K_p\left(\theta_i(t) - \theta_o(t)\right)$$

- Implemented as a multiplier.
- If the difference is small, the mixing approach to the difference.




$$\equiv$$




$$y(t) \approx \frac{1}{2}\sin\left(\theta_i(t) - \theta_o(t)\right) + \text{high frequency terms}$$

$$y(t) = x(t) \times s(t)$$

$$= \cos\left(2\pi f_i t + \theta_i(t)\right) \times -\sin\left(2\pi f_o t + \theta_o(t)\right)$$

$$y(t) = \frac{1}{2}\sin\left(\theta_i(t) - \theta_o(t)\right) - \frac{1}{2}\sin\left(4\pi f_i(t) + \theta_i(t) + \theta_o(t)\right)$$

low frequency term — high frequency term

$$y(t) \approx \frac{1}{2}\left(\theta_i(t) - \theta_o(t)\right)$$

# LOOP FILTER

- The design of the loop filter is vital in defining the overall characteristics and behavior of the PLL.
- The PLL type corresponds to the number of integrators in the loop, including the one that add the VCO/NCO.

| PLL Type | Integrators | | |
|---|---|---|---|
| | VCO/NCO | Loop Filter | Total |
| Type 1 | 1 | 0 | 1 |
| Type 2 | 1 | 1 | 2 |
| Type 3 | 1 | 2 | 3 |

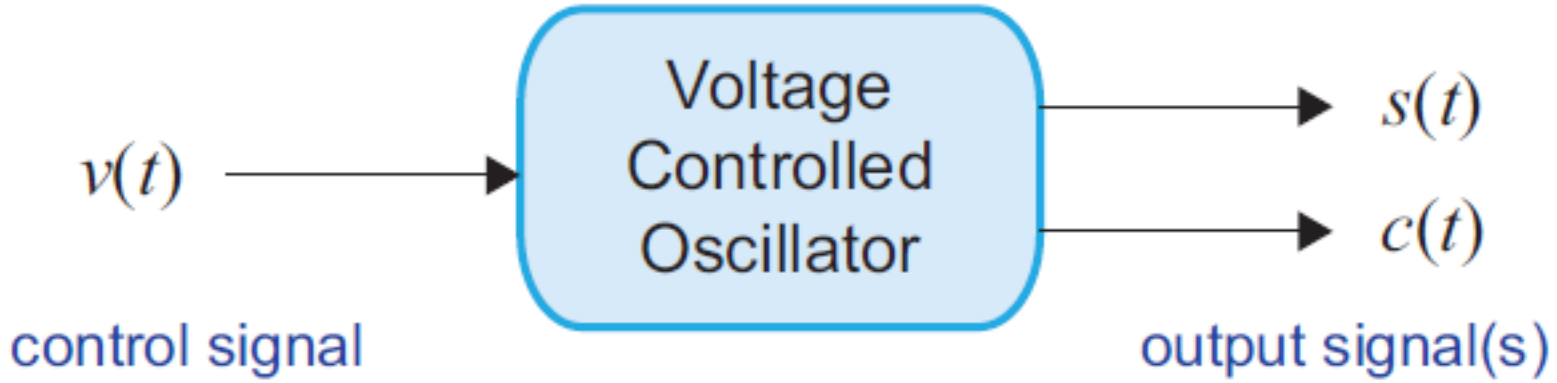
# CONTROLLED OSCILLATOR

- The VCO control signal is the filtered phase difference v(t)
- The estimated phase at instant "t" is :

$$\hat{\theta}(t) = k_o \int_0^t v(t)\, dt$$

- *k0* is the sensibility of VCO

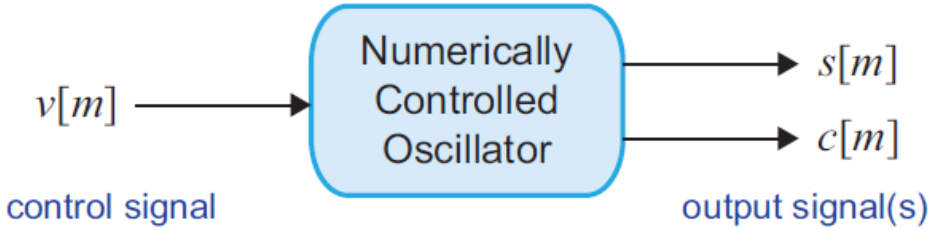
$$c(t) = \cos\left(2\pi f_o t + \hat{\theta}(t)\right)$$



# NCO: NUMERICALLY CONTROLLED OSCILLATORS

- The digital version of VCO.

$$c[m] = \cos\left(2\pi f_o m T + \hat{\theta}[m]\right)$$

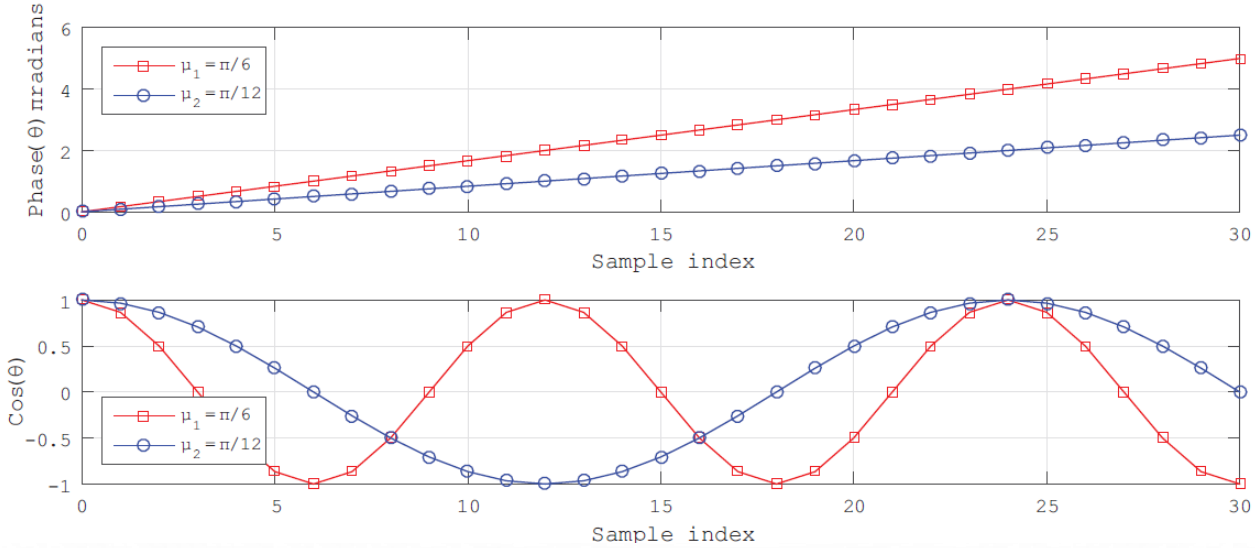



$$\hat{\theta}[m] = K_o \sum_{m=0}^{M} v[m]$$

$$\hat{\theta}[m] = \hat{\theta}[m-1] + \mu[m]$$

$$\mu = \frac{2\pi f_d}{f_s}$$

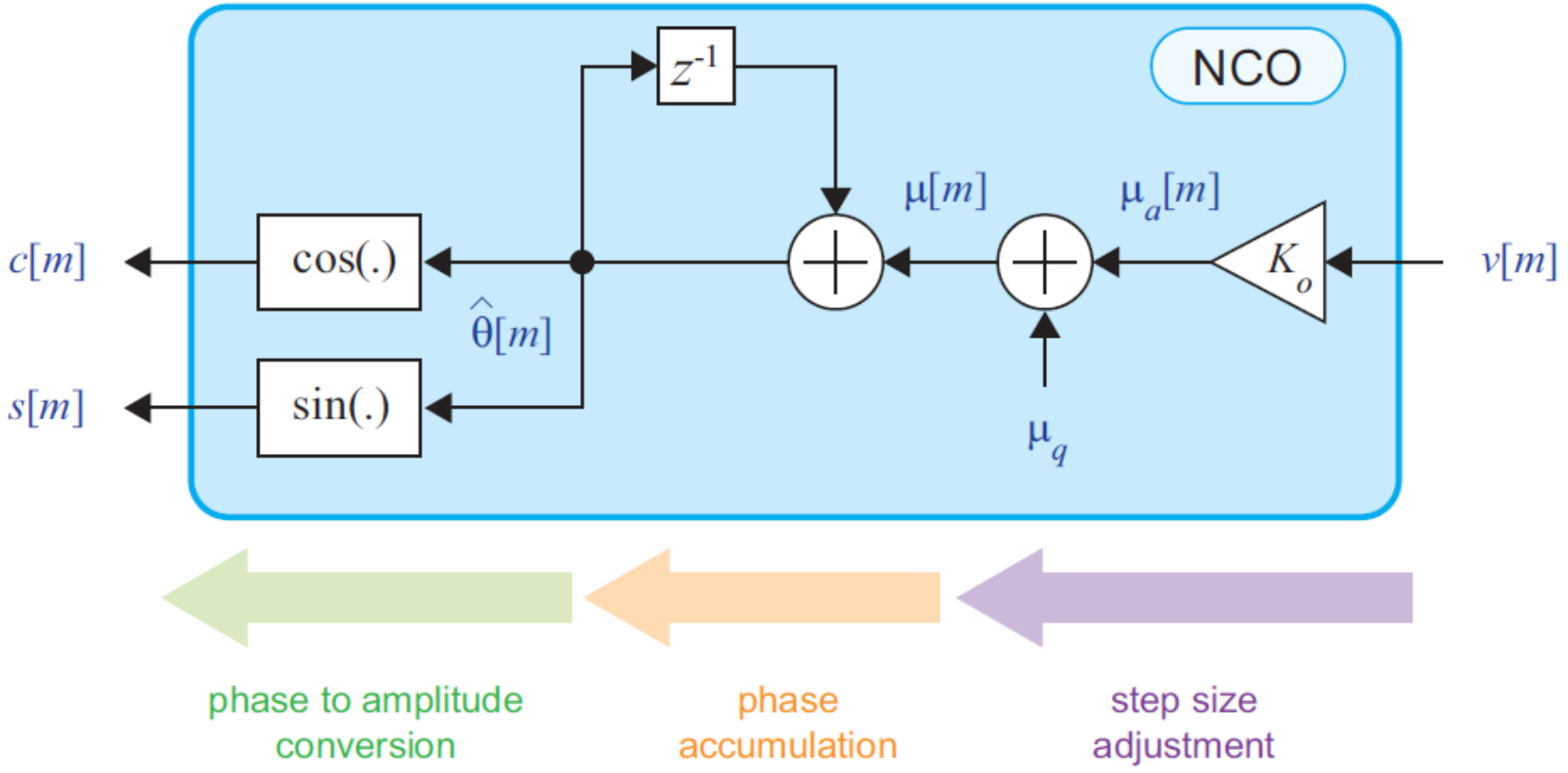
- Different frequencies are created from different step size.



# NCO

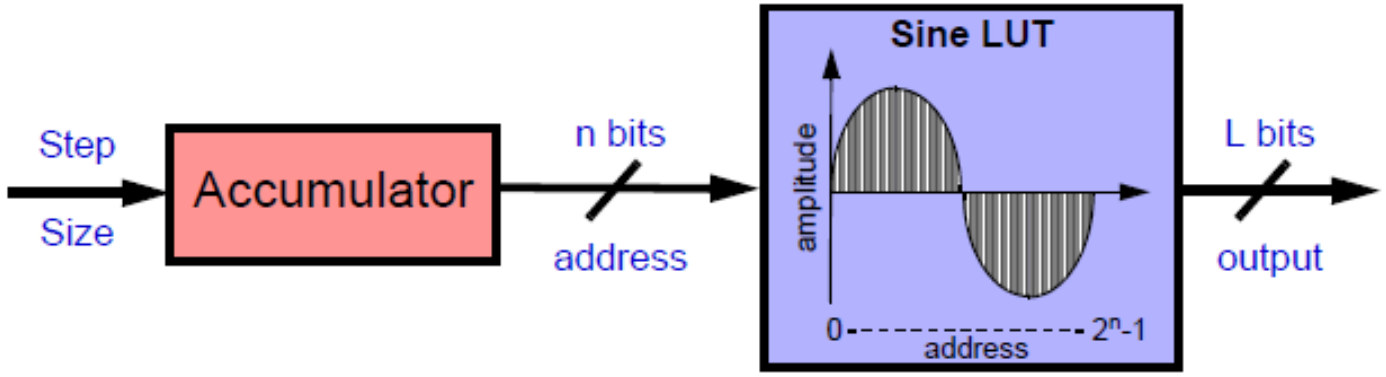
- To follow a frequency change, an adjustment term is needed.

$$\hat{\theta}[m] = \hat{\theta}[m-1] + \mu[m] \qquad \mu[m] = \mu_q + \mu_a[m] \qquad \mu_a[m] = K_o v[m]$$



# NCO: IMPLEMENTATION ON FPGA

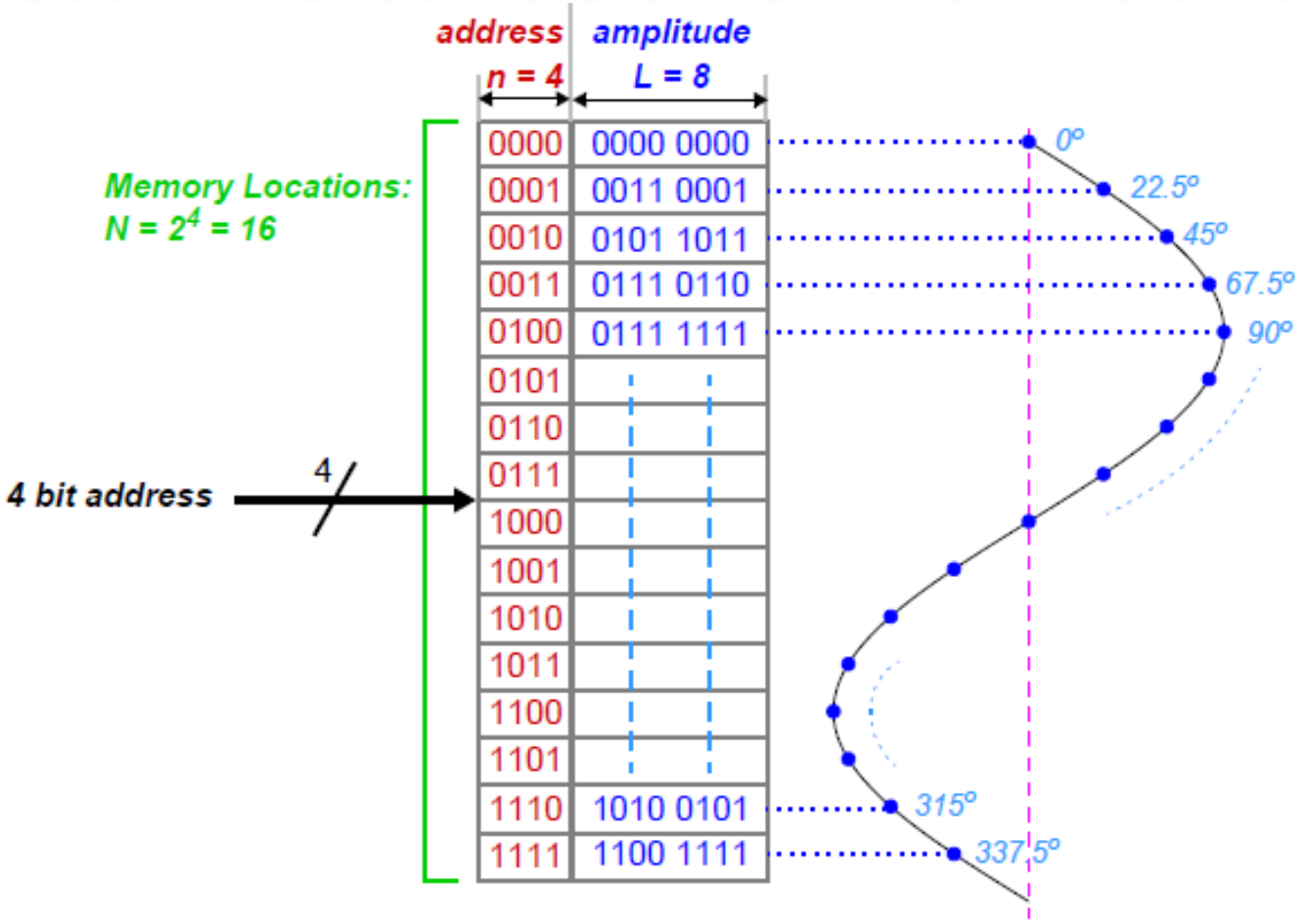
- The simplest operation inside FPGA could be up/down conversion to intermediate frequency.
- NCO are usually implemented on LUTs. Also called Digital Direct Synthesizer (DDS).




- The LUT has $N=2^n$ size, where n is the numbers of bits that accumulator generate.
- The amplitude resolution of the signal depends on the number of **outputs bits L**, and the frequency resolution depends on LUT size, n.

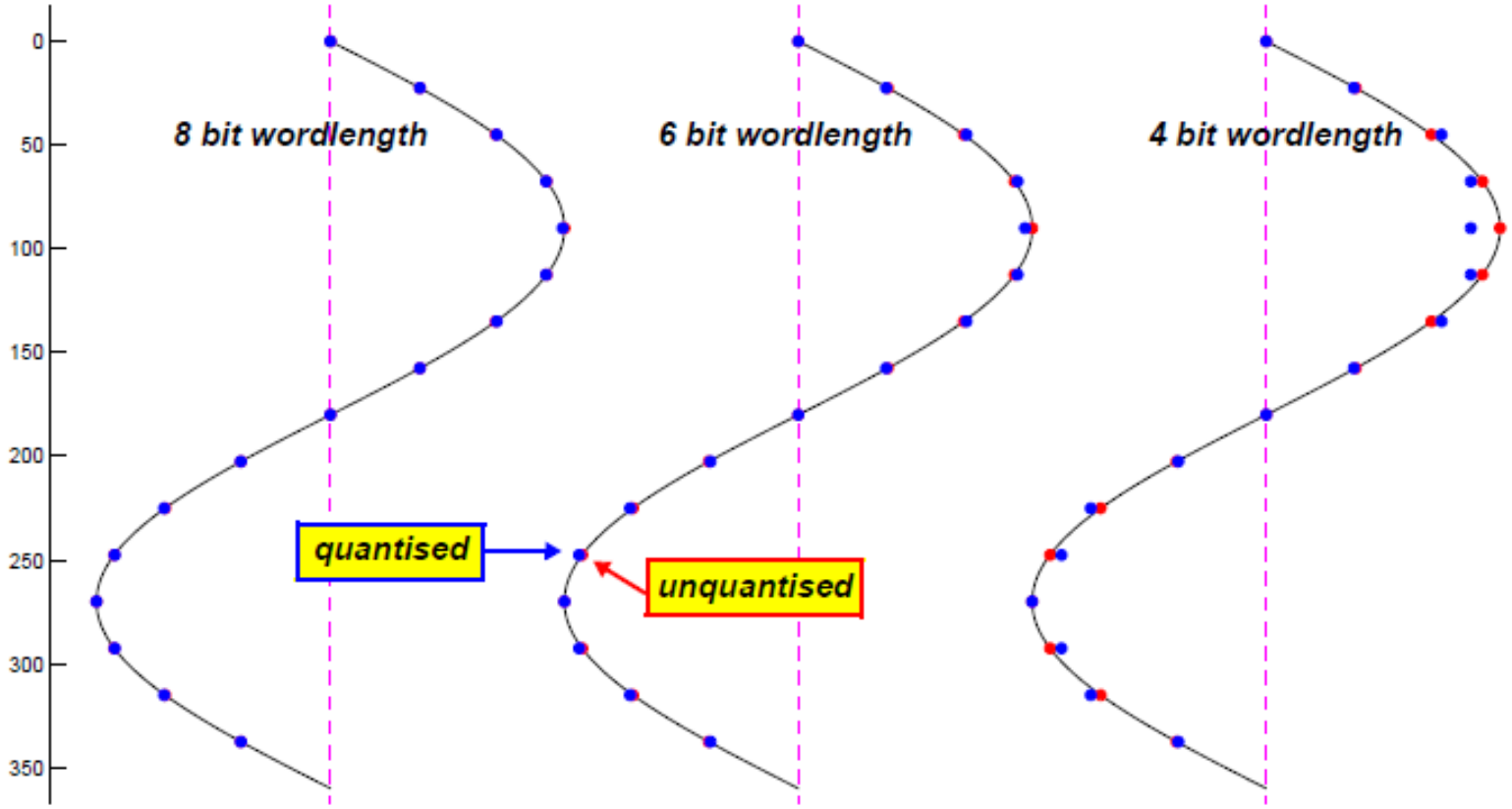
# NCO ON FPGA

- Frequency and amplitude resolution



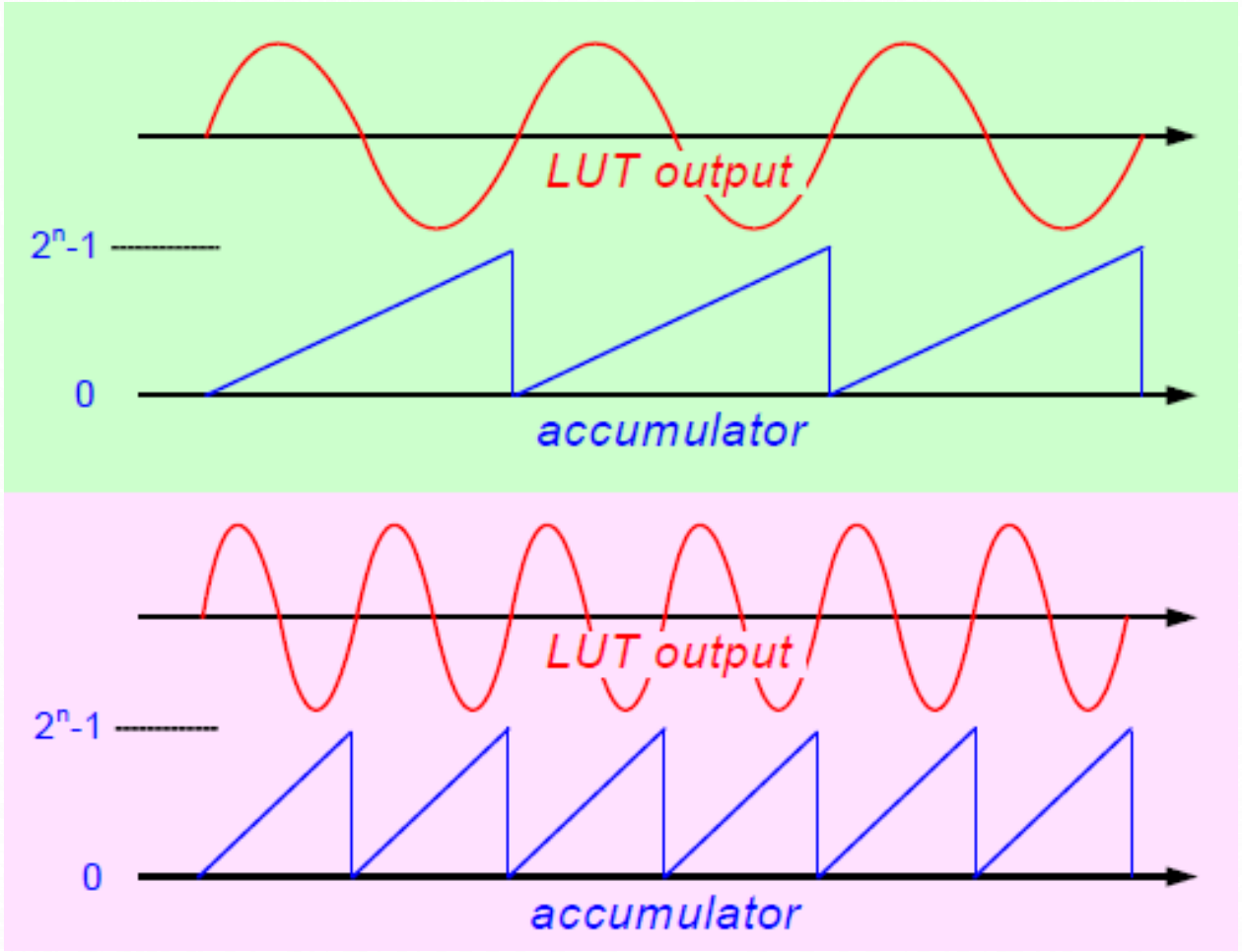
# NCO ON FPGA

- Quantization effect, L



# NCO ON FPGA

- Frequency control is done by the **step** of Accumulator



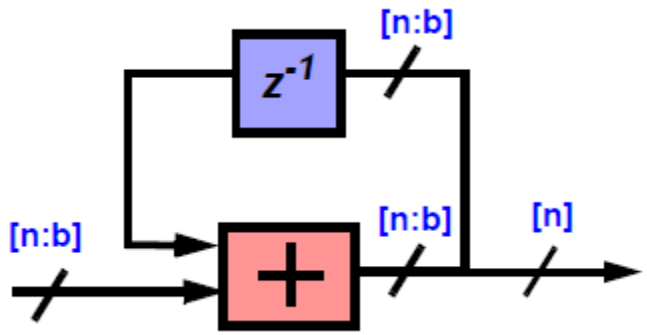
# NCO ON FPGA

- The Accumulator step is determined by:
  - N: number s entries on the LUT.
  - Fs: sampling frequency
  - Fd: desired frequency

$$\mu = N \frac{f_d}{f_s}$$

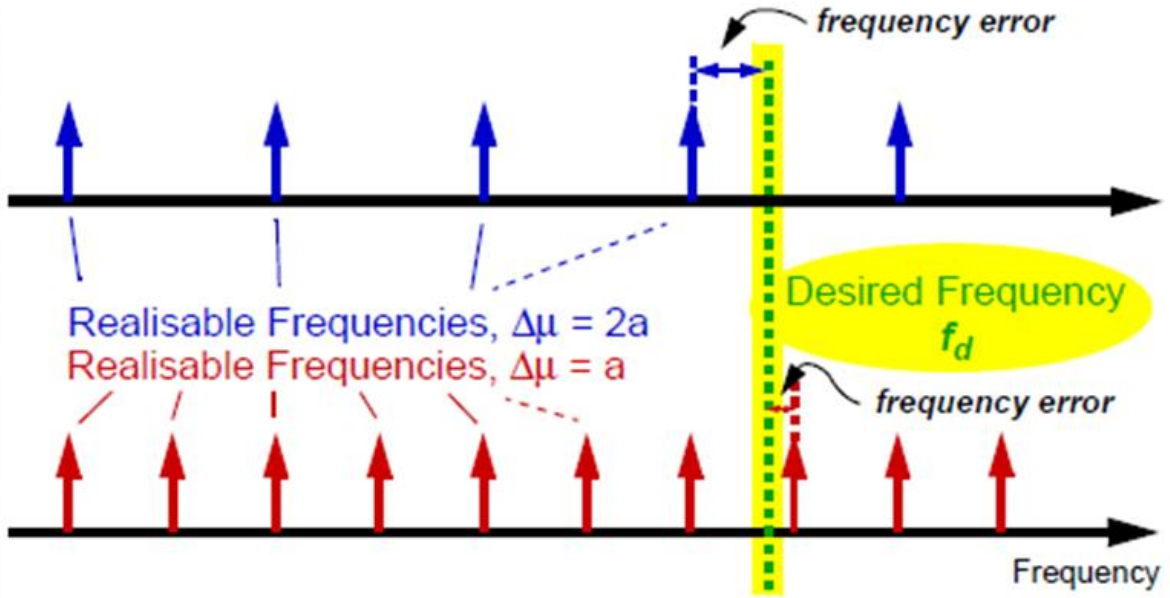
- Example 8 bits, N=256, fs=10Mhz, fd=2.5Mhz $\mu = 256 \frac{2.5MHz}{10MHz} = 64$
- If we need 2.4Mhz?? $\mu = 256 \frac{2.4MHz}{10MHz} = 61.44 \ldots$
- We need to add a fractional part to the ACC.
- The step will have an integer and a fractional part [n:b]
- μ=61.4375 [8:4]

$$\mu \frac{f_s}{N} = 2,399,902Hz$$



# NCO ON FPGA

- Frequency resolution: depends on steps differences. $\Delta\mu = \frac{1}{2^b}$
- The fractional bits should be selected following the desired frequency resolution.



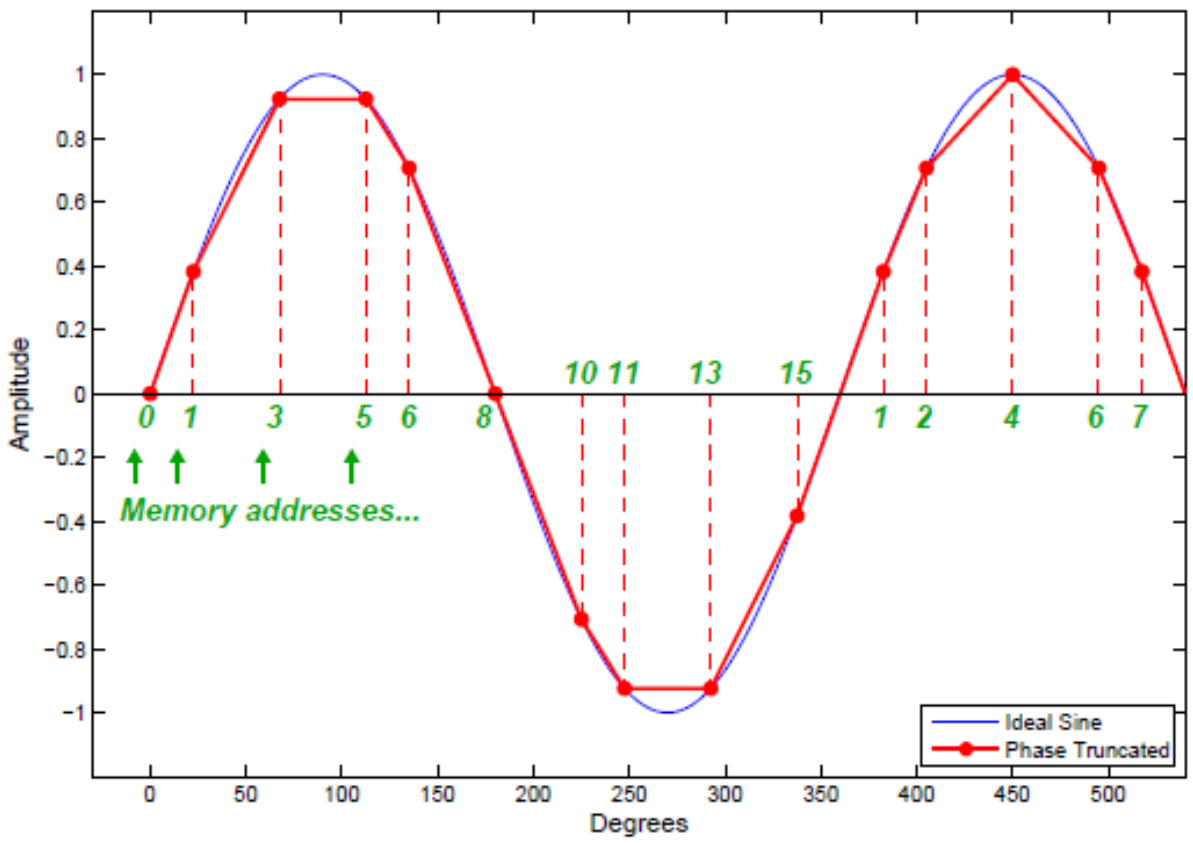

- Frequency resolution $\Delta$fa:

$$\Delta f_a = \Delta\mu \frac{f_s}{N} = \frac{f_s}{2^b N}$$

# NCO ON FPGA

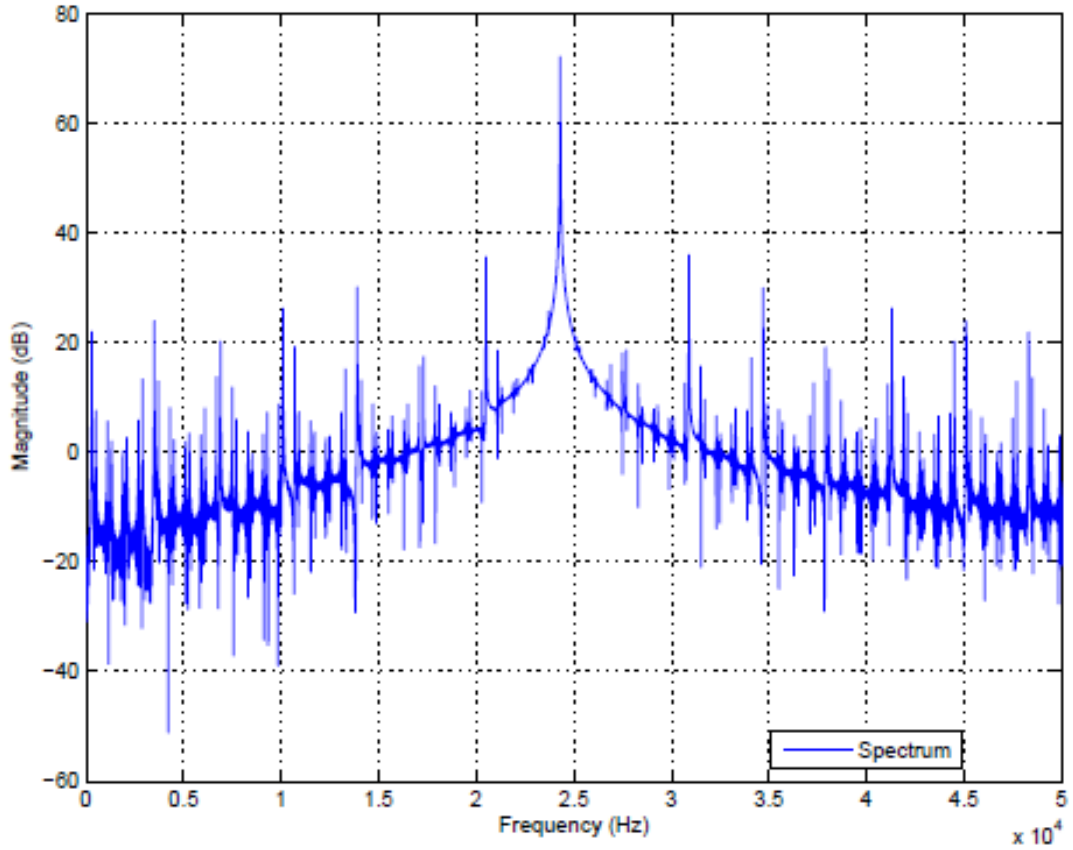
- Truncated error: if the step is μ=1.7, then spurious appear doe to truncate .

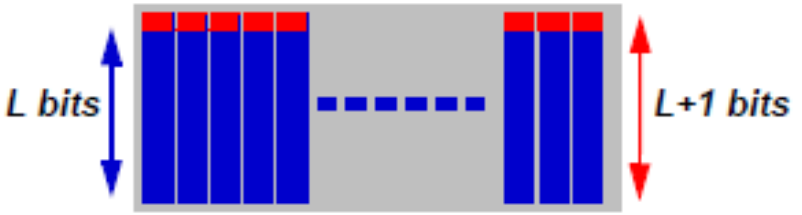
| Sample | Accum. | Address |
|---|---|---|
| 0 | 0 | 0 |
| 1 | 1.7 | 1 |
| 2 | 3.4 | 3 |
| 3 | 5.1 | 5 |
| 4 | 6.8 | 6 |
| 5 | 8.5 | 8 |
| 6 | 10.2 | 10 |
| 7 | 11.9 | 11 |
| 8 | 13.6 | 13 |
| 9 | 15.3 | 15 |
| 10 | 1.0 | 1 |
| 11 | 2.7 | 2 |
| 12 | 4.4 | 4 |
| 13 | 6.1 | 6 |
| 14 | 7.8 | 7 |



# NCO ON FPGA

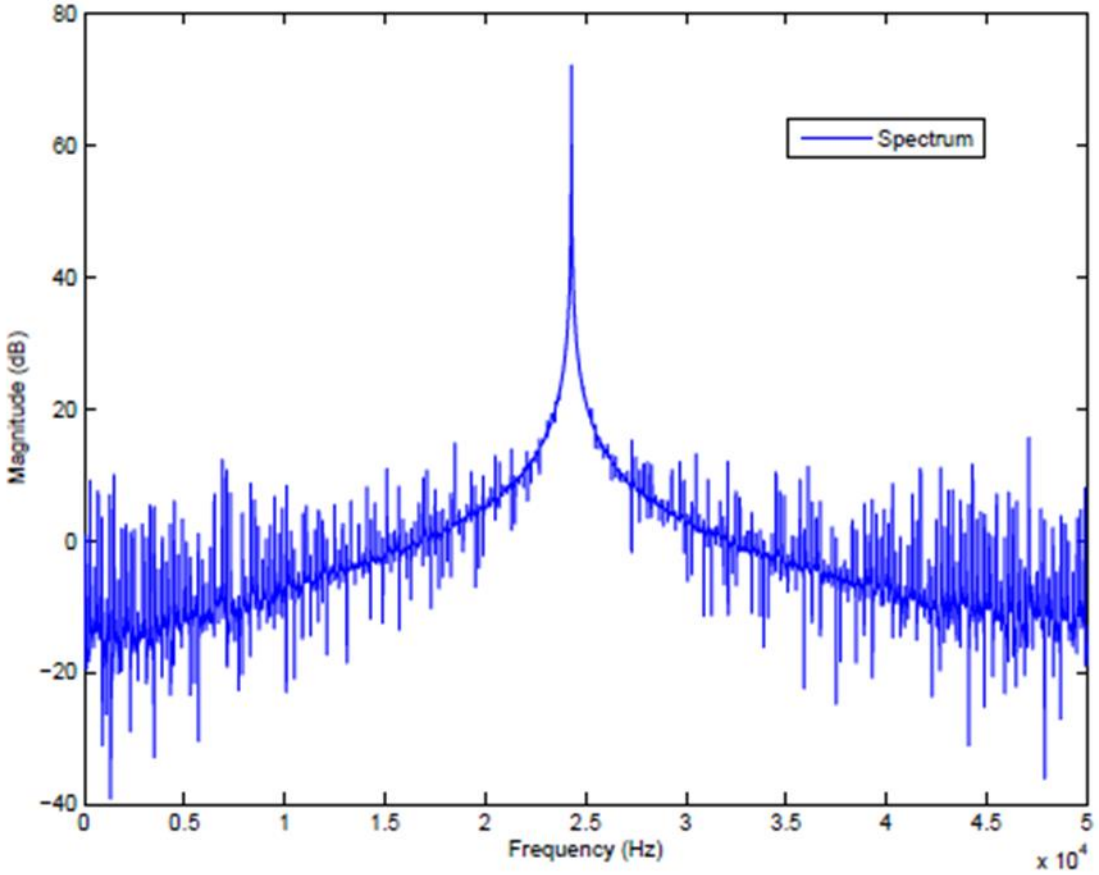
- TRUNCATED ERROR due to small N.
- Fs: 100kHz, fd: 24.3kHz, N:6, [n:b]=[**6**:16], **L=32**

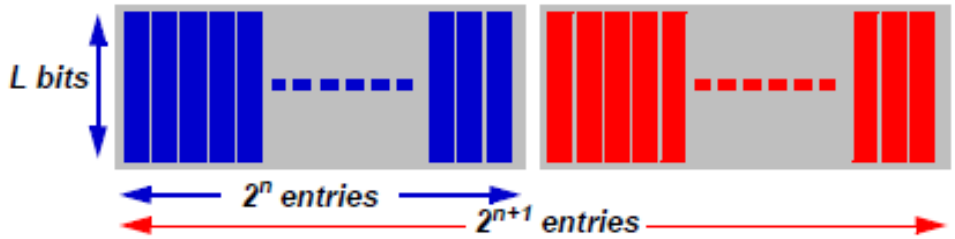


# NCO ON FPGA

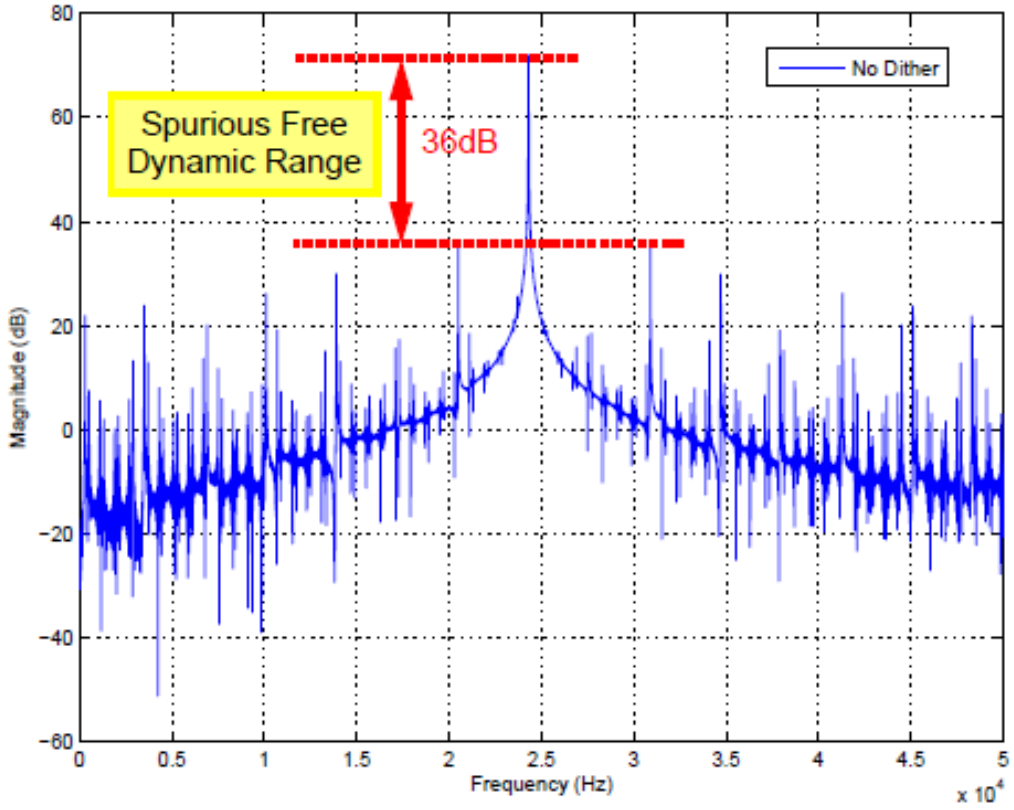
- QAUNTIZATION ERROR, small L
- fs: 100kHz, fd: 24.3kHz, **L:8,** [n:b]=[**12**:16]



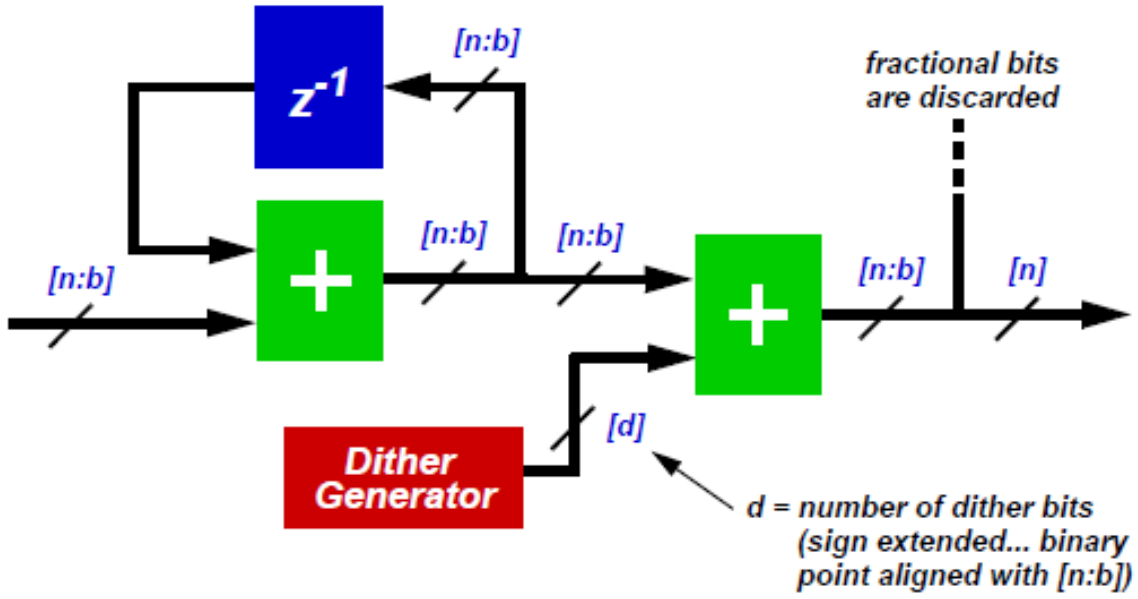
# NCO ON FPGA

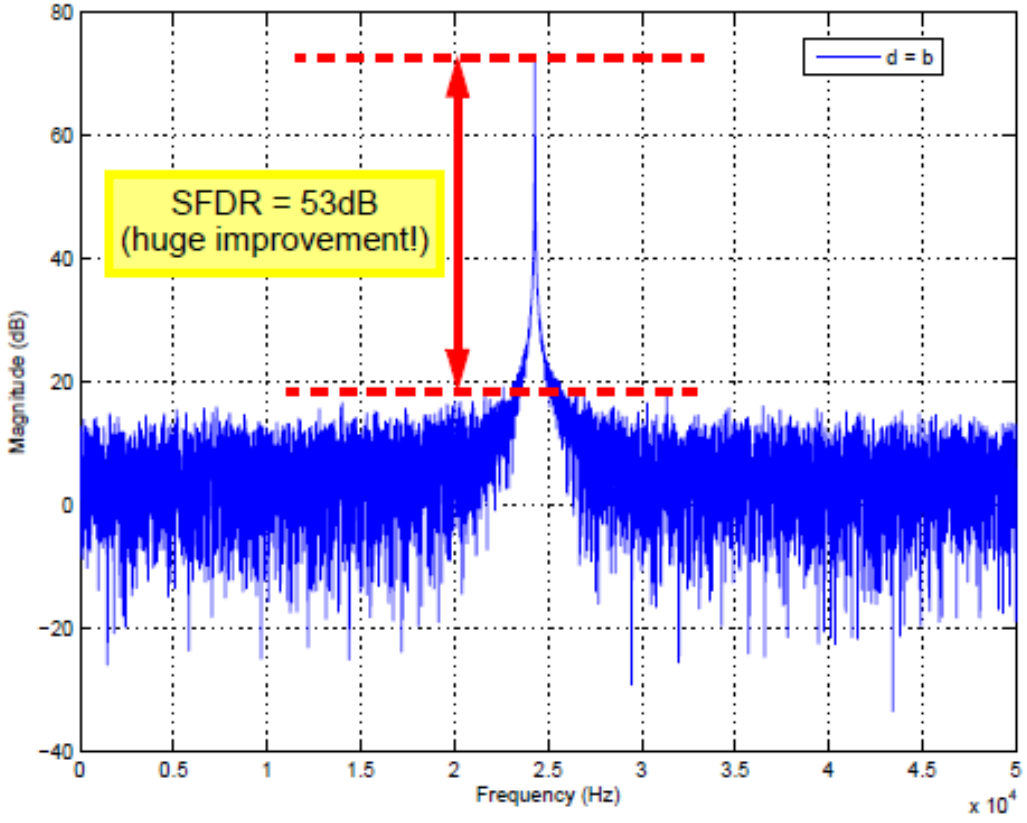
- SFDR: Range free of spurious frequencies
- GSM requires 110Db of SFDR



# NCO ON FPGA

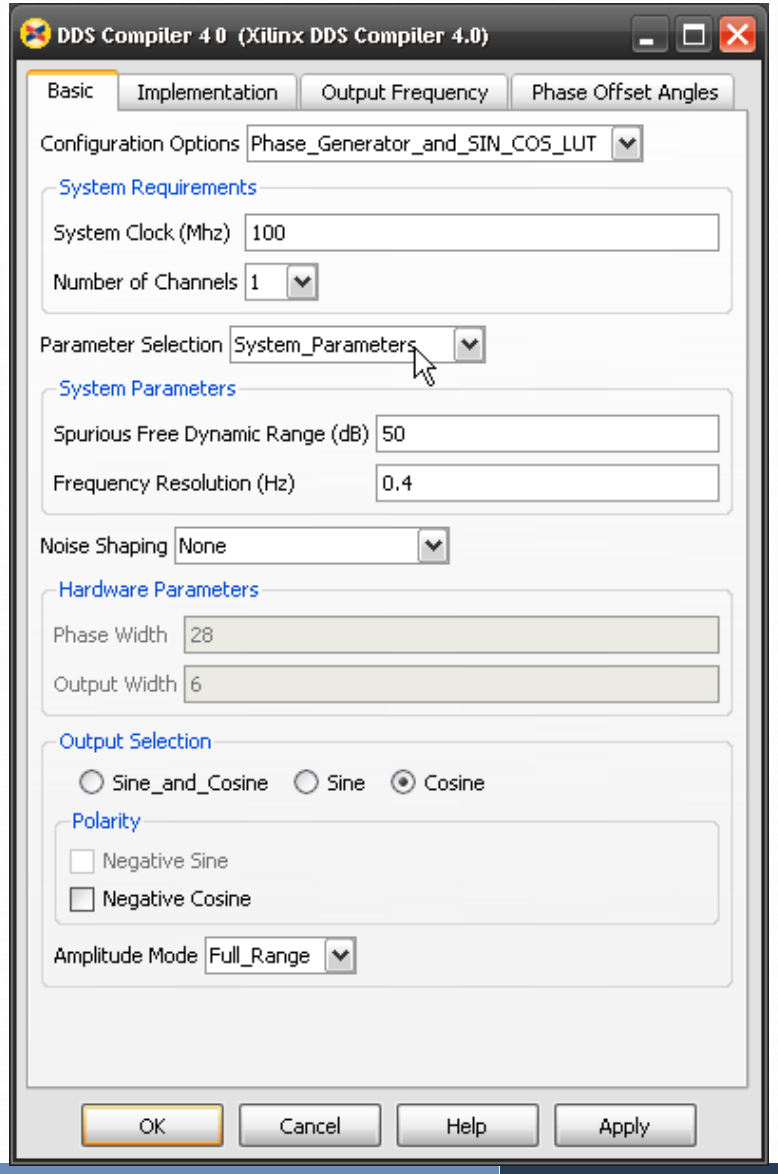
- Increase LUT size N help, but **cost a lot.**
- Better solution: add a dither signal to **break the quantization error.**
- Usually the number of dither bits is equal to fractional bits, **b=d**



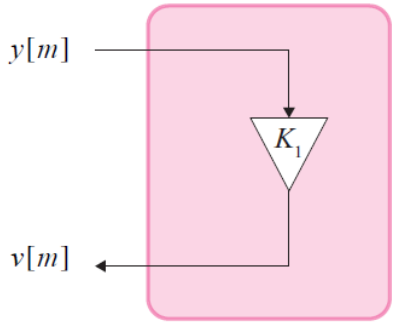
# NCO ON FPGA

- DIRECT DIGITAL SYNTHESIZER

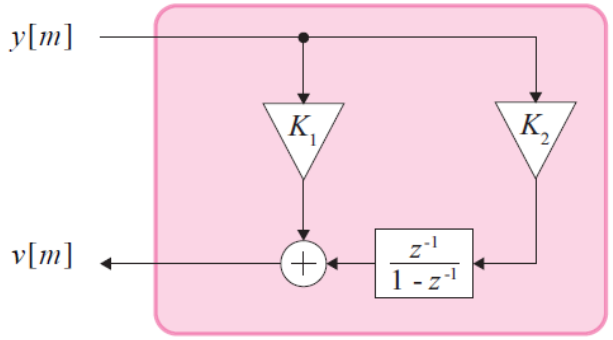


# LOOP FILTERS TYPES

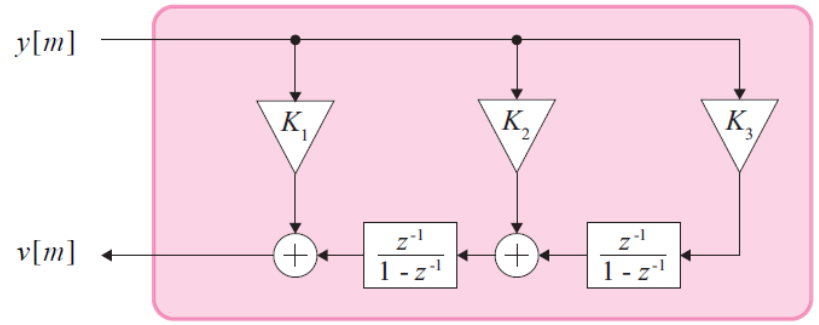
*Type 1 Loop Filter*

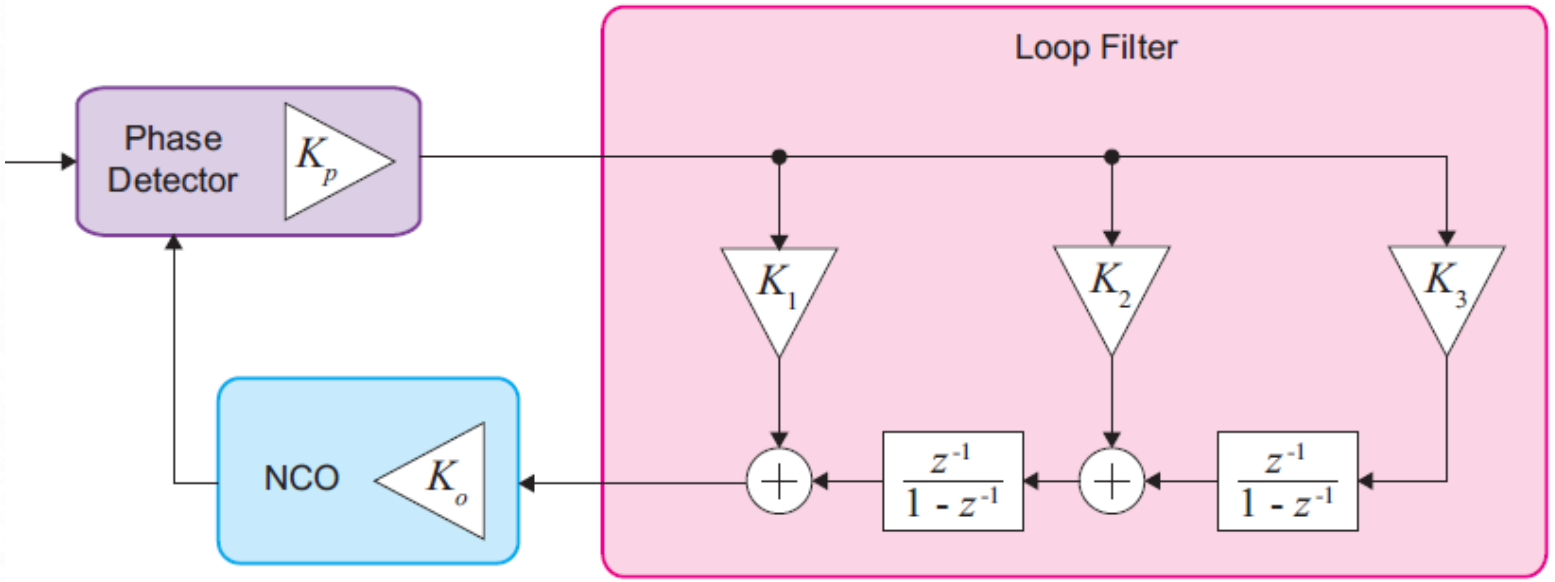



*Type 2 Loop Filter*



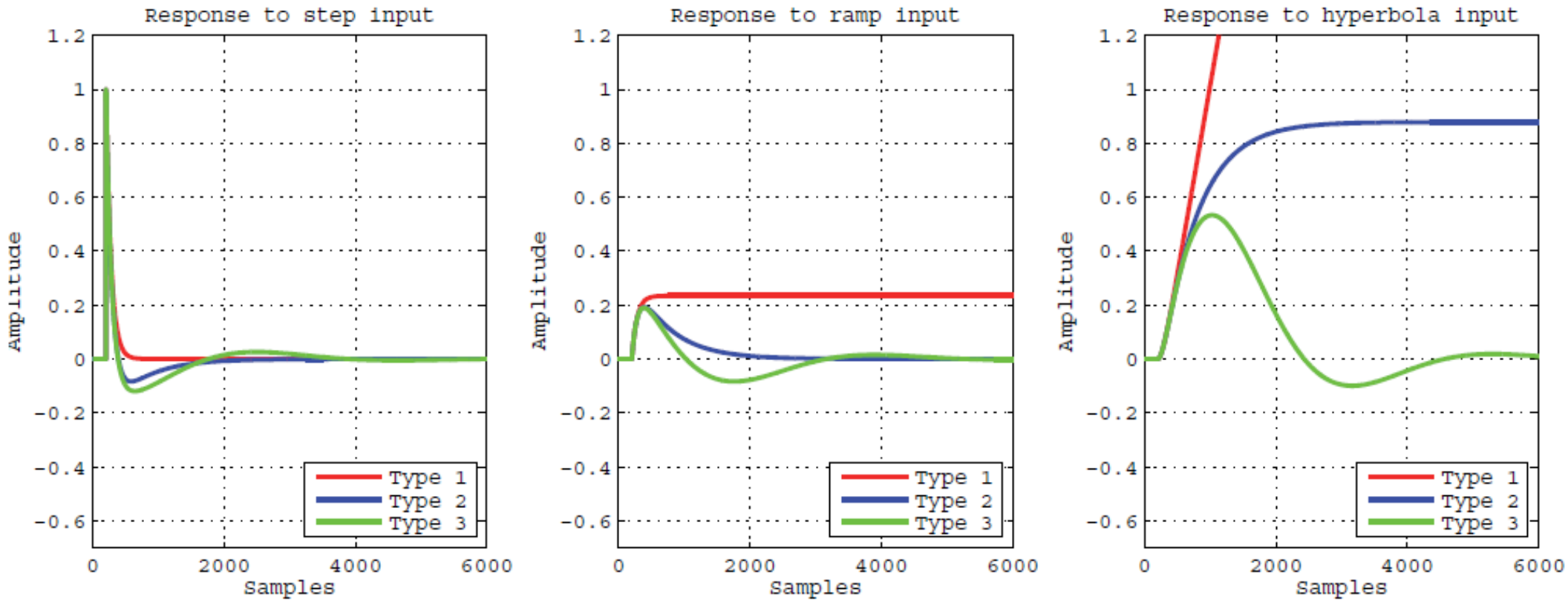

*Type 3 Loop Filter*



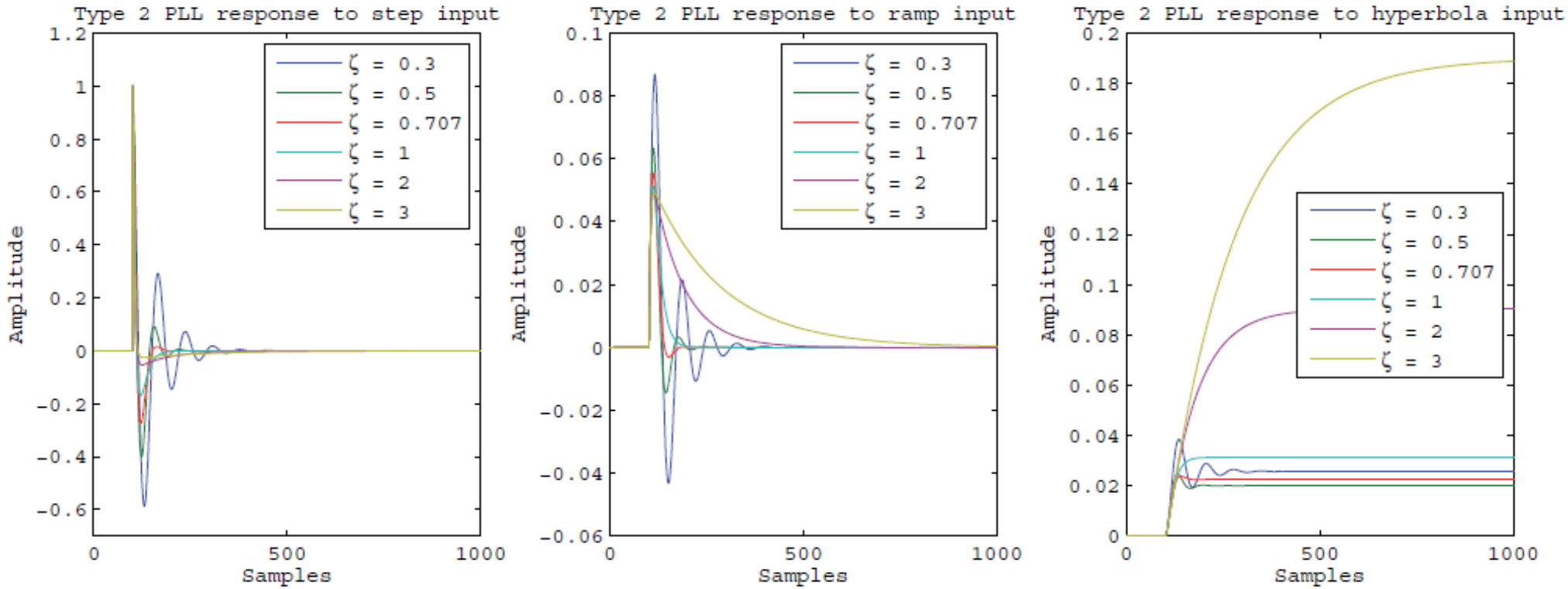
# DESIGN PARAMETERS

- Time to achieve lock, depends on the step size.
- Steady state error, depends on the number of integrators and how the input signal change.
- Tracking capabilities, deepens on the PLL type.



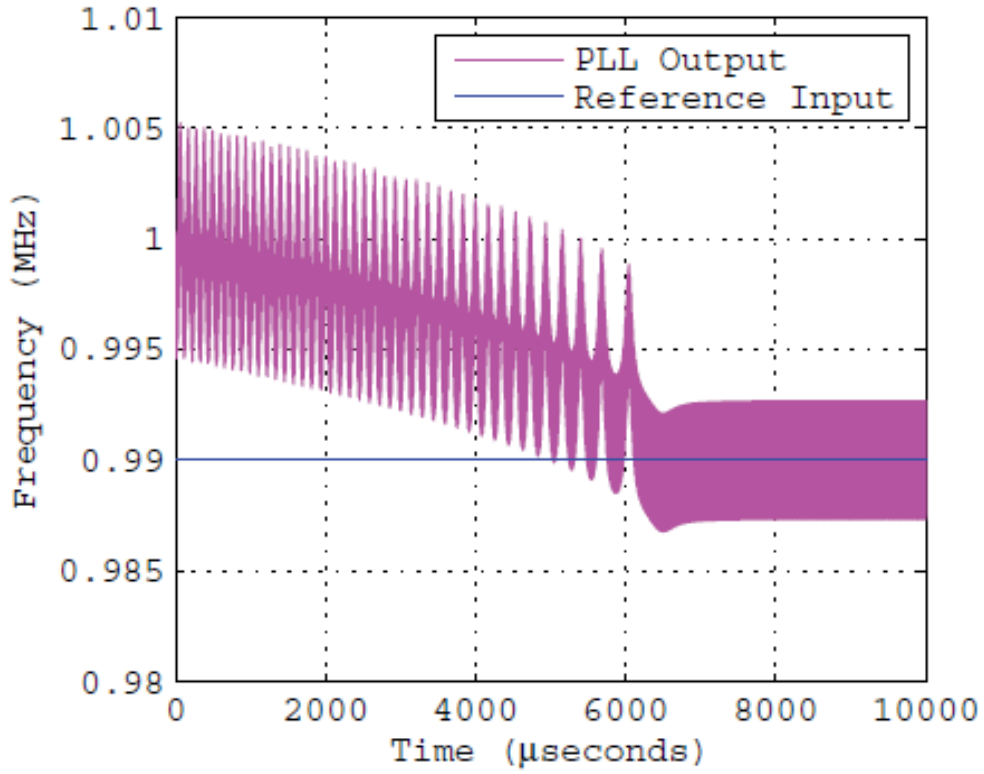
# DAMPING RATIO

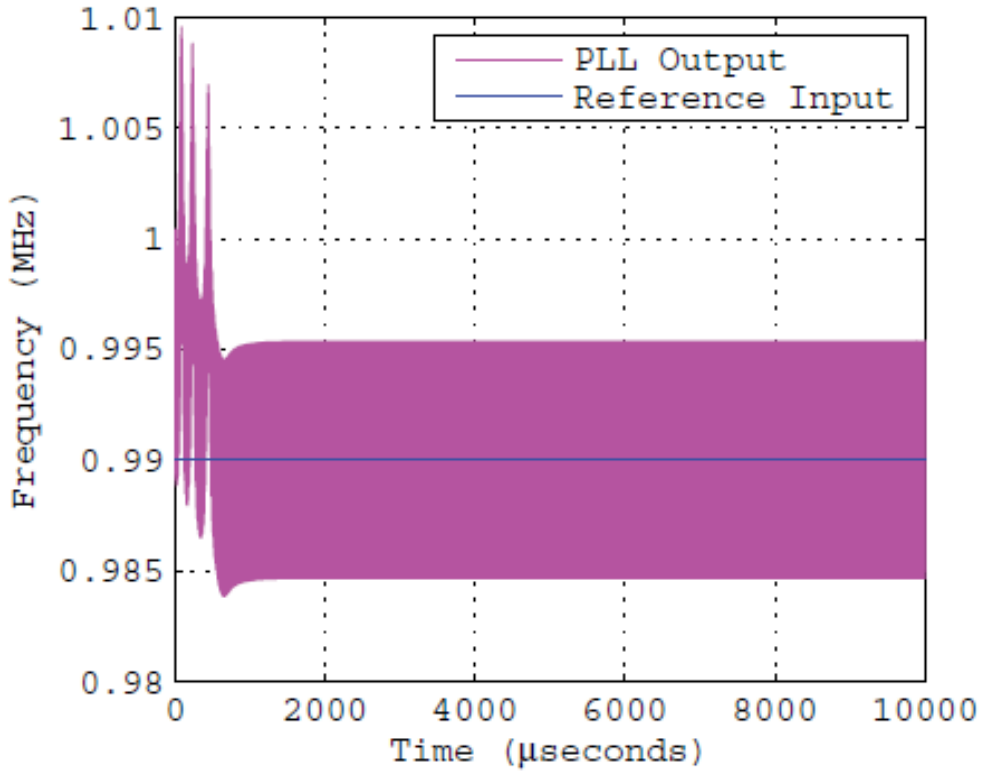
- $\zeta < 1$ under-damped $\zeta > 1$ over damped $\zeta = 1$ critically damped
- The damping factor, or damping *ratio*, relates to the transient behavior of the PLL as it achieves phase lock.
- Typical value is 0,707



# BANDWIDTH

- The bandwidth refers to the range of frequencies over which the PLL operates
- At lower BW, bigger transient time at bigger BW lower transient. Cons: more noise into the PLL.

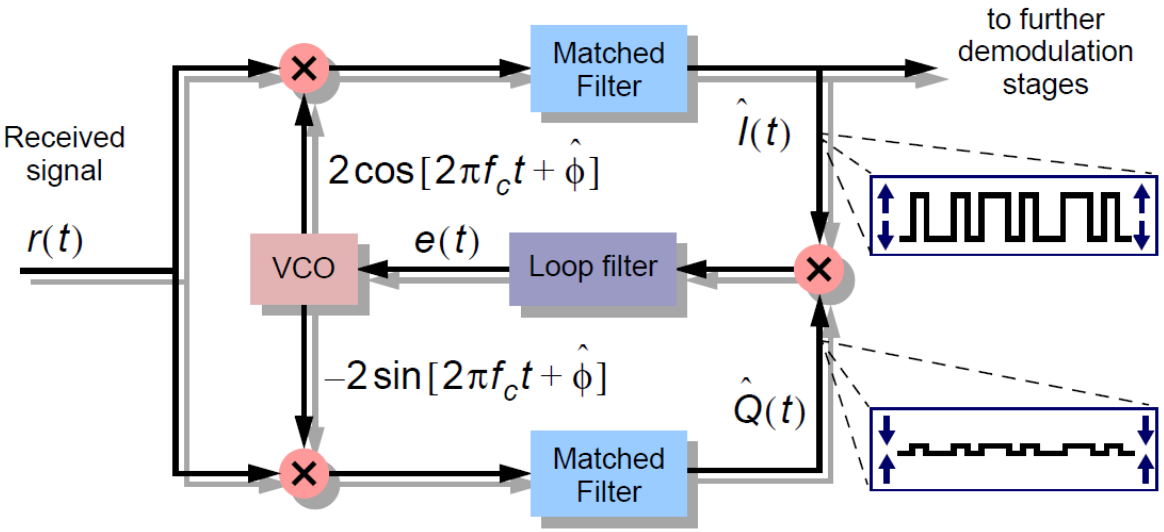


# COSTAS LOOP

- It is a type of PLL used on AM-DSB-SC demodulation. Also used on M-PSK demodulations.
- It is based on the sin vs cos orthogonality.
- Principal advantage is its double sensibility . **sen(2($\theta i - \theta f$))**
- Especially useful for Doppler effect correction.

$$r(t) = A(t)\cos(2\pi f_c t + \phi)$$




$$\hat{I}(t) = A(t)\cos(\phi - \hat{\phi})$$

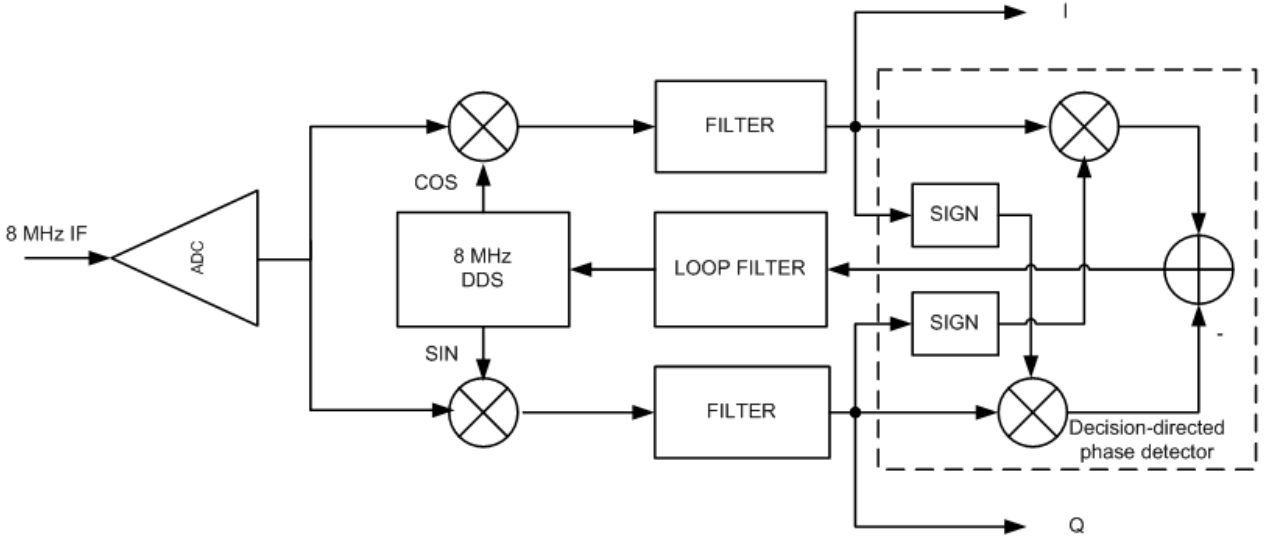
$$\hat{Q}(t) = A(t)\sin(\phi - \hat{\phi})$$

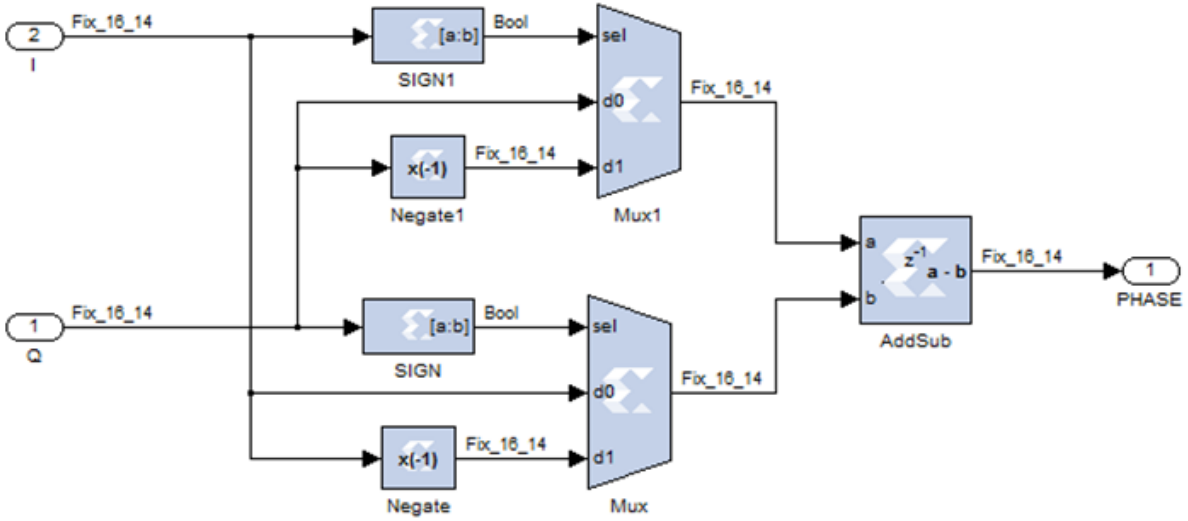
$$e(t) = A^2(t)\cos(\phi - \hat{\phi})\sin(\phi - \hat{\phi})$$

$$= \frac{1}{2}A^2(t)\sin(2(\phi - \hat{\phi}))$$

# COSTAS LOOP FPGA IMPLEMENTATION

- Decision Direct PLL



# COSTAS LOOP EXAMPLE

- 4QAM RX/TX