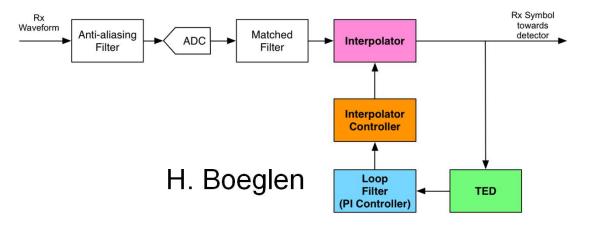


From a simulated to a real digital communication system: a QPSK modern design with GNU Radio



1

XLIM, UMR CNRS 6172, Université de Poitiers, France

European GNU Radio Days 2023, Paris, France

Outline

- 1 Introduction and motivation
- 2 QPSK modem: the main building blocks
- 3 The fully digital QPSK synchronizer.
- 4 GNU Radio demo with real hardware
- 5 Conclusion

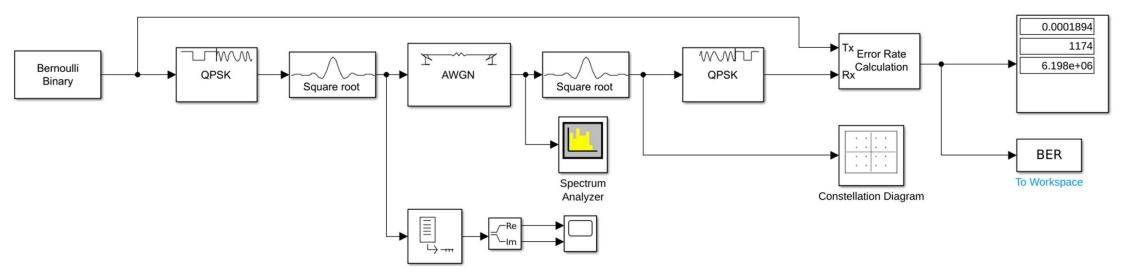
Outline

- 1 Introduction and motivation
- 2 3 – 4 – 5 –

1. Introduction & motivation

□ I have been teaching digital communications (DC) at undergraduate and graduate levels for more than 15 years.

Extensive use of Matlab/Simulink and the Communications toolbox:



→ Very good tool to illustrate the main DC concepts by simulation.

□ Up to this year in the 2nd year of the IoT master's degree two disjoint topics teached in the DC course:

Basics + ECC (me)
 Hardware aspects: linearity in RF amplifiers (my collegue)

Everything is teached using Matlab/Simulink simulation tools.

□ My collegue has discovered the power of GNU Radio in a DPD (digital pre-distorsion) research project.

□ We have decided to give more coherence to the course by:

Replacing Matlab/Simulink by GNU Radio as much as possible (at last; I do not want to have trouble with JM ⁽ⁱ⁾).

1. Introduction & motivation

- → Giving more space to implementation with the design of a real world QPSK modem (GNU Radio + hardware (Adalm-Pluto) up to the linearized PA).
- □ So this presentation will be about the difficulties you'll face when you go from simulation to implementation!
- □ Since it is the topic of these 2 days we will focus mainly on synchronization aspects.
- □ Well, no choice: no synchronization means no real world implementation anyway.



Outline

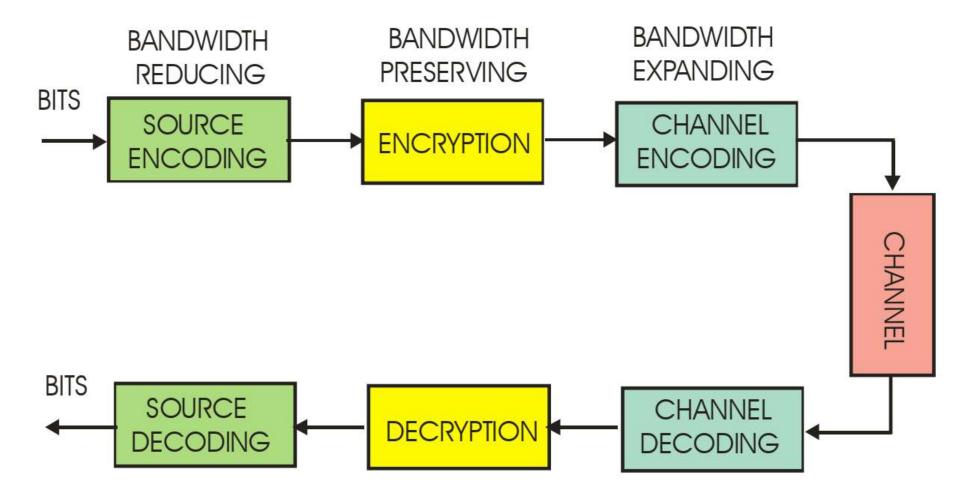
2 – QPSK modem: the main building blocks

7

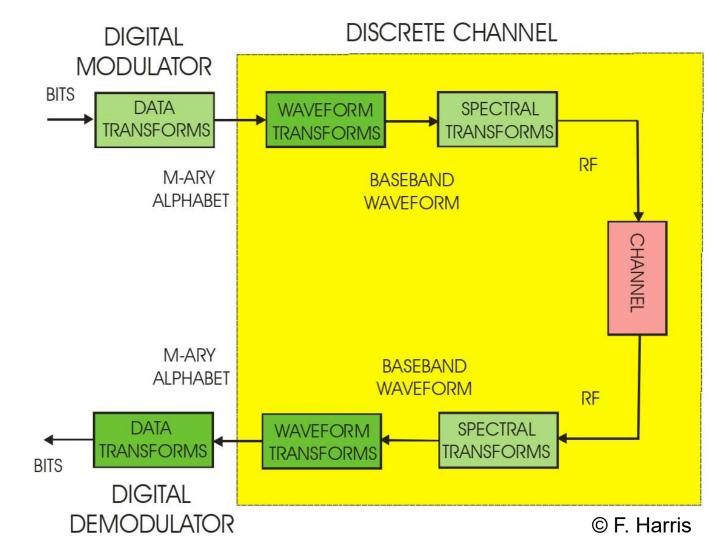
3 – 4 – 5 –

1 -

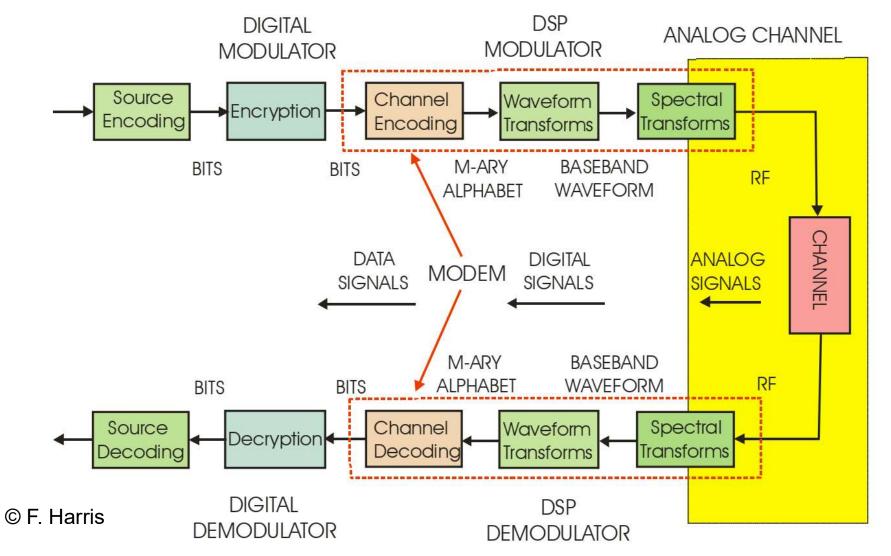
□ From Shannon to the digital QPSK modem



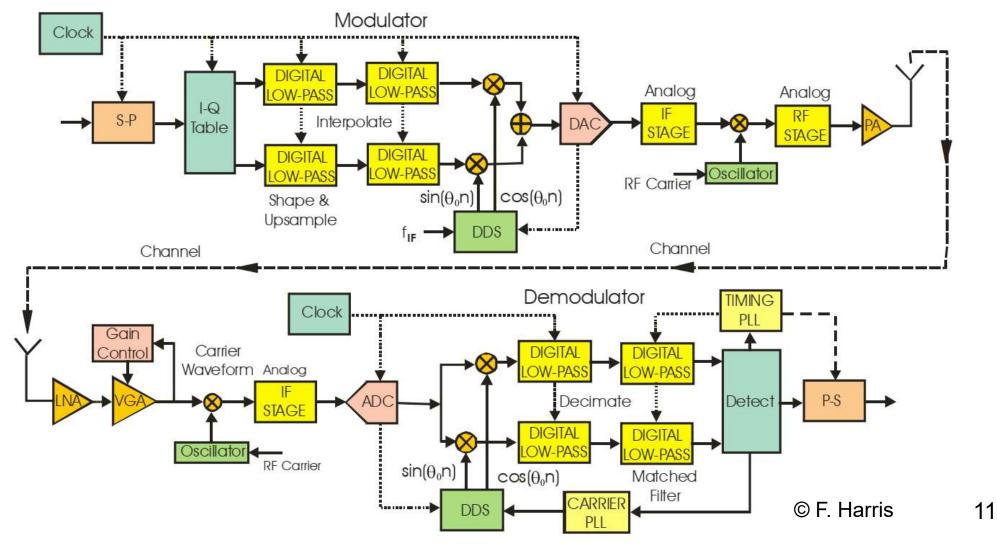
□ From Shannon to the digital QPSK modem



□ From Shannon to the digital QPSK modem



□ From Shannon to the digital QPSK modem



□ Pulse shaping and matched filtering needed for:

→ Limiting the BW
→ Maximizing SNR at the decision points
→ Reduce ISI

❑ Use of the well known Root Raised Cosine (RRC) pulse
 → One at the transmitter and one at the receiver = Raised Cosine response, i.e. Nyquist ISI criterion.

□ In practice: what are the important parameters?

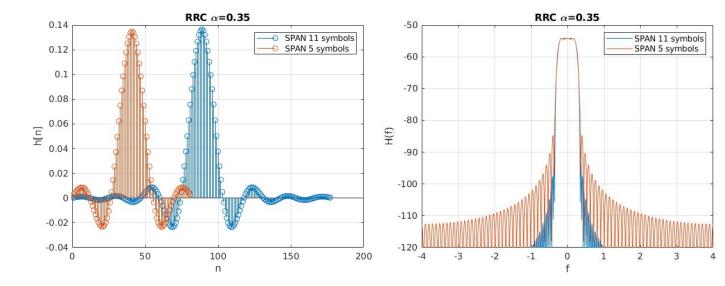
→ Upsampling factor N (SPS): (4 to 16)

→ Filter span (number of symbols).

 \rightarrow Roll-off α (excess bandwith/cardinal sine): between 0.25 and 0.6.

□ RRC pulse filter coefficients generation

- →Matlab h = rcosdesign(α, SPAN, SPS)
- →Octave h = rcosfir(α,nT,SPS,T,'sqrt')
- →GNU Radio firdes.root_raised_cosine(gain, SPS,T, α,ntaps (SPAN*SPS))
- RRC pulse filter example ($\alpha = 0.35$, SPS = 16, SPAN = 11)



firdes.root_raised_cosine(2, 2*8, 1.0, 0.35, (11 or 5)*8*2)

□ How many samples per symbol (N)?

N = Ts/T with Ts = sample rate and T = symbol rate

In practice: $4 \le N \le 16$

□ Be careful about the signal level at the reception side!

- At baseband: system with symbol of unit energy.
- Amplitude A is then given by:

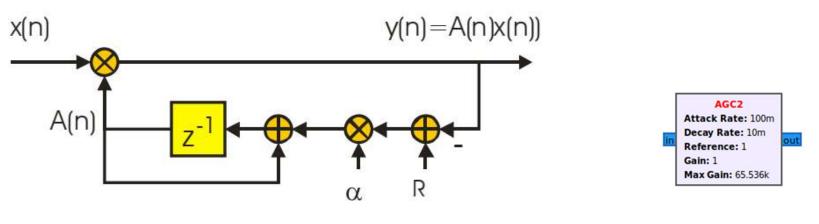
→A = sqrt((3*Es)/(2*(M-1)) with Es = symbol energy in J and M = number of points of the constellation.

→ For QPSK A = sqrt(2.0) = 0.707.

Receiver functions designed to handle this value which has to remain constant → need for an AGC

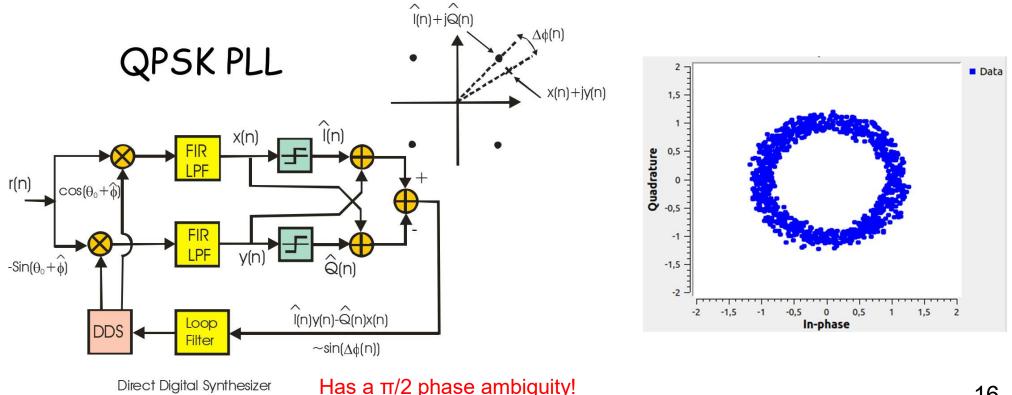
□ Signal level at the reception side: AGC

- Need to ensure a constant signal level for the receiver function to work
- Signal cannot be constant because propagation channel varies
- → Use of an Automatic Gain Control (AGC) structure

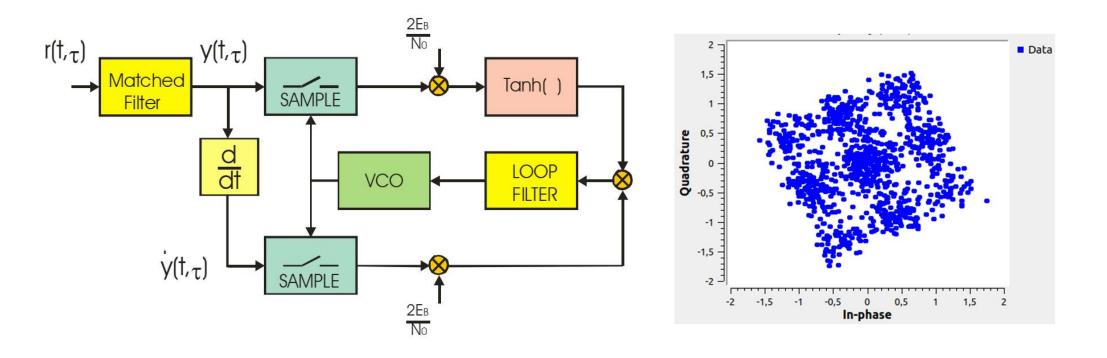


 $\begin{array}{l} A[n+1] = A[n](1-\alpha c) + \alpha R \\ R = reference, \ \alpha = gain \ c = constant \ and \ \alpha c < 2.0. \\ Loop \ time \ constant \ is \ 1/\alpha c \ samples. \\ If \ c \ small \ long \ transient. \ If \ c \ large \ short \ transient. \end{array}$

- □ Some more loops: carrier and timing synchronization PLLs
- **Carrier and phase recovery loop**
- → TX oscillator unknown phase
- \rightarrow Generated carrier frequency f_{Rx} at receiver is not exactly f_{Tx}

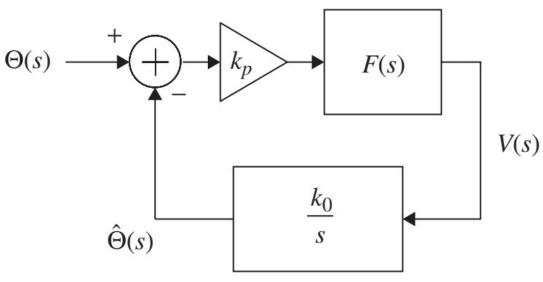


- □ Some more loops: carrier and timing synchronization PLLs
- Timing recovery loop
- → Optimum sampling instants have to be extracted from the data



□ From analog PLL to digital PLL and its main parameters

- We will restrict to a second order loop (the most used).
- The analog PLL



Proportional-plus- Transferment Transferment

Transfer function:

$$F(s) = k_1 + \frac{k_2}{s} \qquad H_a(s) = \frac{k_0 k_p k_1 s + k_0 k_p k_2}{s^2 + k_0 k_p k_1 s + k_0 k_p k_2}$$

Can be rewritten:

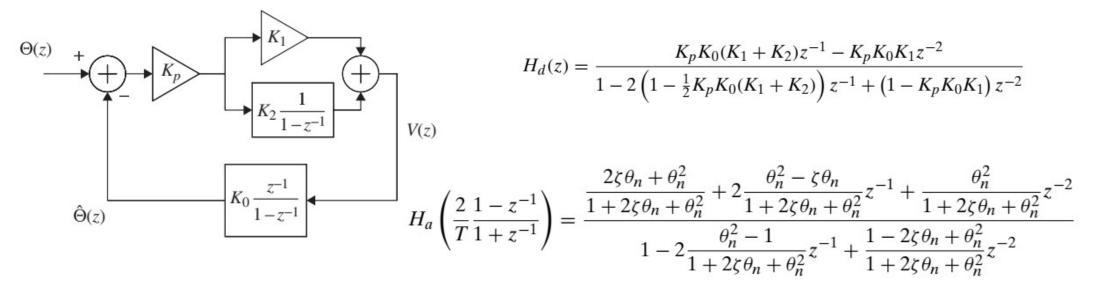
$$H_a(s) = \frac{2\varsigma\omega_n s + \omega_n^2}{s^2 + 2\varsigma\omega_n s + \omega_n^2}$$

With:

PLL BW:
$$\omega_{3dB} = \omega_n \sqrt{1 + 2\varsigma^2 + \sqrt{(1 + 2\varsigma^2)^2 + 1}}$$
 $\varsigma = \frac{k_1}{2} \sqrt{\frac{k_0 k_p}{k_2}}$ $\omega_n = \sqrt{k_0 k_p k_2}$
PLL noise BW: $B_n = \frac{\omega_n}{2} \left(\varsigma + \frac{1}{4\varsigma}\right)$

From analog PLL to digital PLL and its parameters

- We will restrict to a second order loop (the most used).
- The digital PLL (apply Tustin's equation)

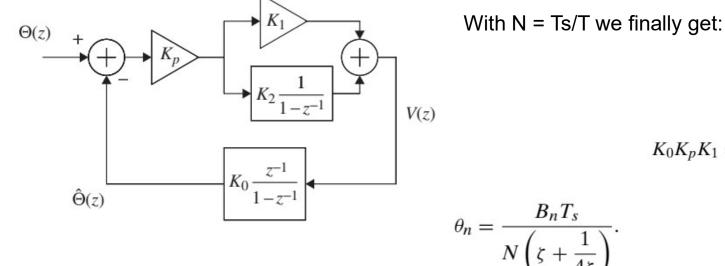


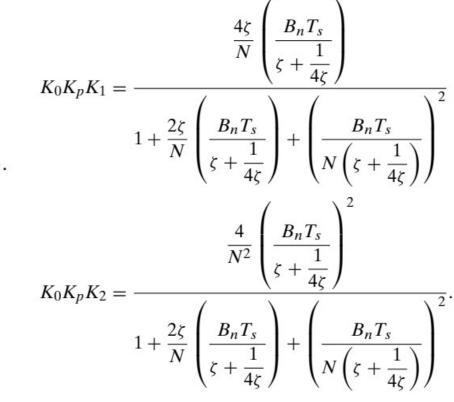
Equating the denominator polynomials in Hd and Ha gives :

$$K_p K_0 K_1 = \frac{4\zeta \theta_n}{1 + 2\zeta \theta_n + \theta_n^2}$$
$$K_p K_0 K_2 = \frac{4\theta_n^2}{1 + 2\zeta \theta_n + \theta_n^2}.$$
 19

From analog PLL to digital PLL and its parameters

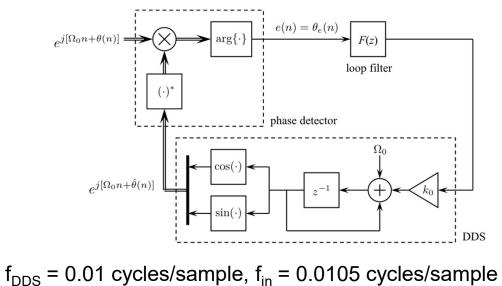
- We will restrict to a second order loop (the most used).
- The digital PLL (apply Tustin's equation)



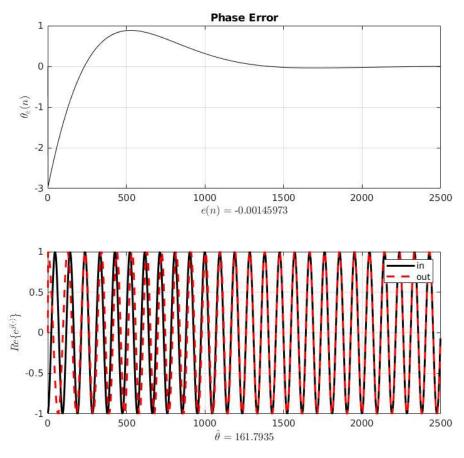


From analog PLL to digital PLL and its parameters
 Digital PLL example

Digital PLL example



 $\zeta = 0.707, B_n T = 0.2\%$



We get: K1 = 0.0053 and K2 = 1.4184e-05

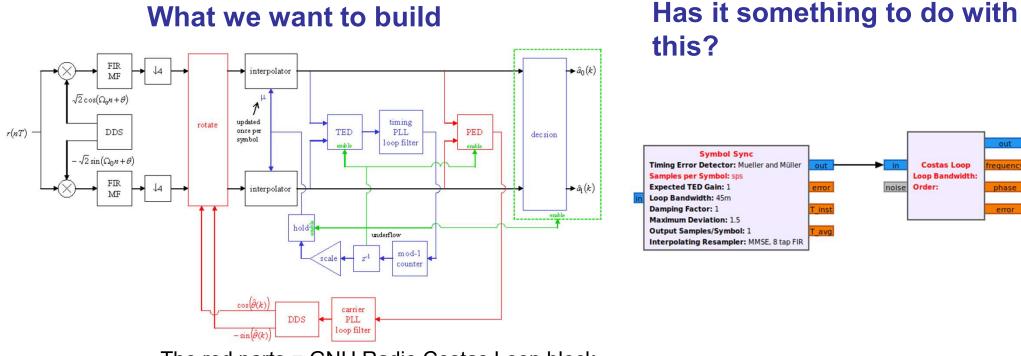


- 2 -2 - The fully digital OPSK synchronizer
- 3 The fully digital QPSK synchronizer.
- 4 -

1 -

5 -

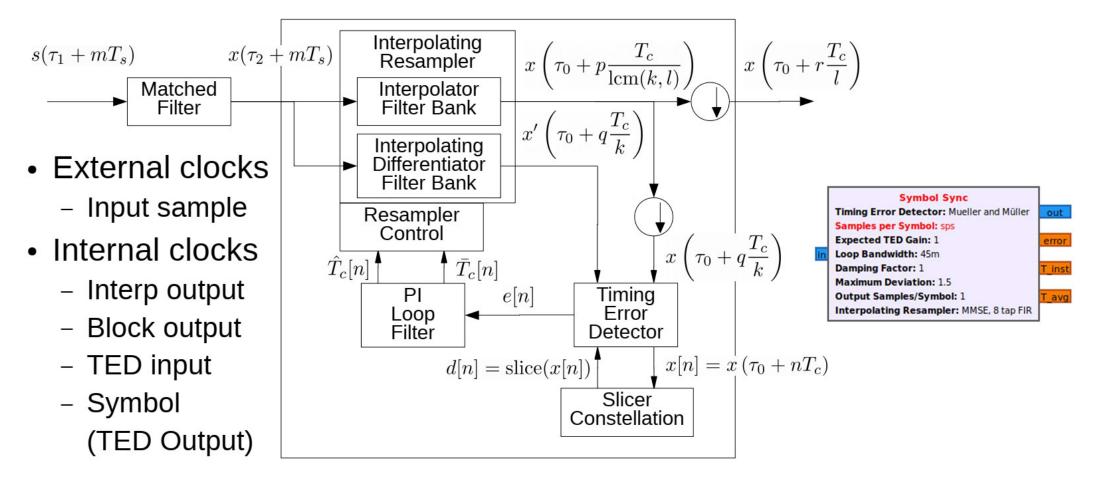
□ The all digital phase/frequency and timing synchronizer



The red parts = GNU Radio Costas Loop block

□ Let's concentrate on the symbol timing synchronization

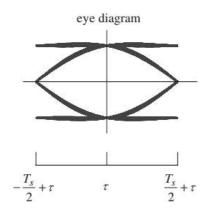
□ The timing synchronizer and A. Wall's GNU Radio Symbol Sync block

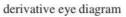


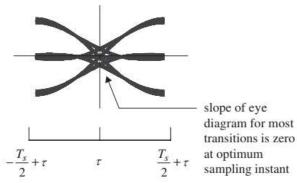
→ Two new elements: Interpolator and Timing Error Detector (TED)

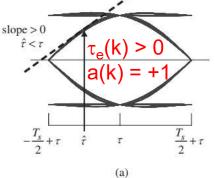
□ The timing synchronizer: Timing Error Detector (TED)

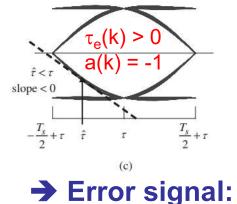
- Plays the role of phase detector in the DPLL
- Best understood with the eye diagram whose slope can be used to generate a timing error e(k) (binary PAM)



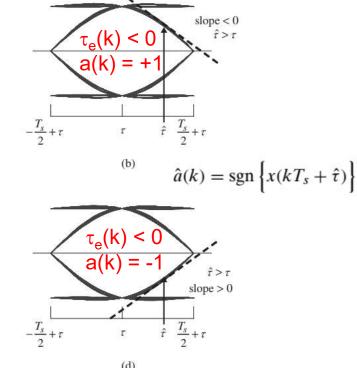


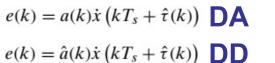










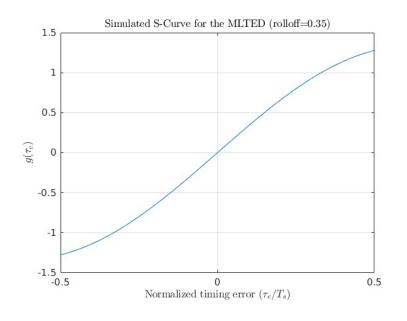


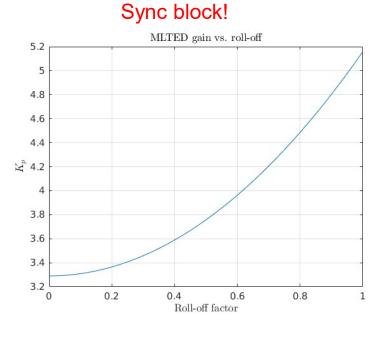
□ The timing synchronizer: Timing Error Detector (TED)

- There exists quite a lot of TED structures (ZC, M&M, Gardner etc.).
- Characteristics of TED described by the S-curve $g(\tau_e)$
- S-curve obtained by computing expected value of error signal:

$$g(\tau_e) = \mathrm{E}\{e(k)\}$$

• For example for the MLTED we get:

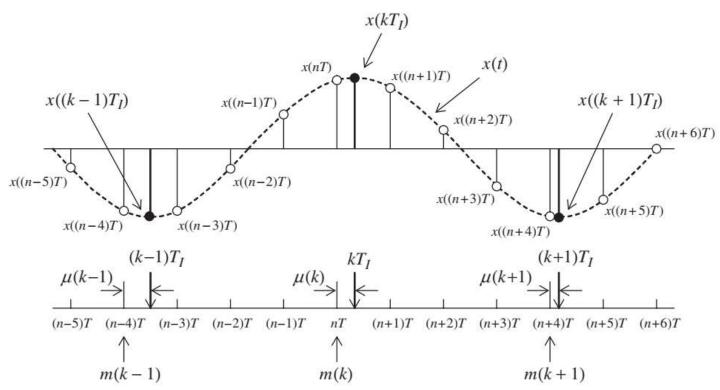




Needed by Symbol

□ The timing synchronizer: Interpolator structures

- Because we use asynchronous sampling (i.e. sample clock independent to the data clock used at the transmitter) we need an interpolator.
- Some vocabulary:

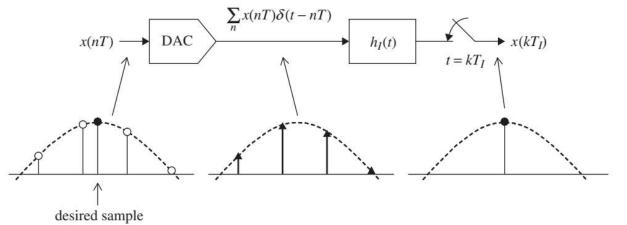


- k-th interpolant = kT₁
 - When k-th *interpolant* is between samples x(nT) and (x(n + 1)T) the sample index n is called the k-th *basepoint index* and denoted m(k).
 - kT₁ is some fraction of a sample time greater than m(k)T.
- This fraction is called the kth fractional interval and is denoted μ(k). 0 < μ(k) < 1.
- It is defined by:

 $\mu(\mathbf{k})\mathbf{T} = \mathbf{k}\mathbf{T}_{\mathbf{I}} - \mathbf{m}(\mathbf{k})\mathbf{T}$

□ The timing synchronizer: Interpolator structures

• Conceptually, an interpolator can be seen as a filter.



• The k-th interpolant evaluated at t = kT₁ and may be expressed as:

$$x(kT_I) = \sum_n x(nT)h_I(kT_i - nT).$$

If we reexpress with a filter index i, m(k) = int(kT_I/T) and μ(k) = kT_I/T- m(k) the filter index is i = m(k) - n. Then:

$$x(kT_I) = \sum_{i} x\left((m(k) - i) T\right) h_I\left((i + \mu(k)) T\right).$$
28

- □ The timing synchronizer: Interpolator structures
- From now to build the interpolator we can:
 - → Use a FIR piecewise polynomial filter

➔ A polyphase-filterbank (massively upsample input of matched filter then downsample matched filter output at an appropriately chosen sample offset to get desired interpolant).

- Let's have a look at the first possibility.
- The continuous underlying waveform is approximated by a polynomial of the form:

$$x(t) \approx c_p t^p + c_{p-1} t^{p-1} + \dots + c_1 t + c_0.$$

 The polynomial coefficient are determined by the p + 1 sample values surrounding the basepoint index. Once the coefficients values are known, the interpolant at t = kTI = (m(k) + μ(k)T is obtained by:

$$x(kT_I) \approx c_p(kT_I)^p + c_{p-1}(kT_I)^{p-1} + \dots + c_1(kT_I) + c_0.$$

□ The timing synchronizer: Interpolator structures

- Let's have a look at the first possibility.
- It can be shown that:

$$x((m(k) + \mu(k))T) = \sum_{i=-2}^{1} h_2(i)x((m(k) - i)T)$$

• In the case of a parabolic interpolator this leads to four coefficients:

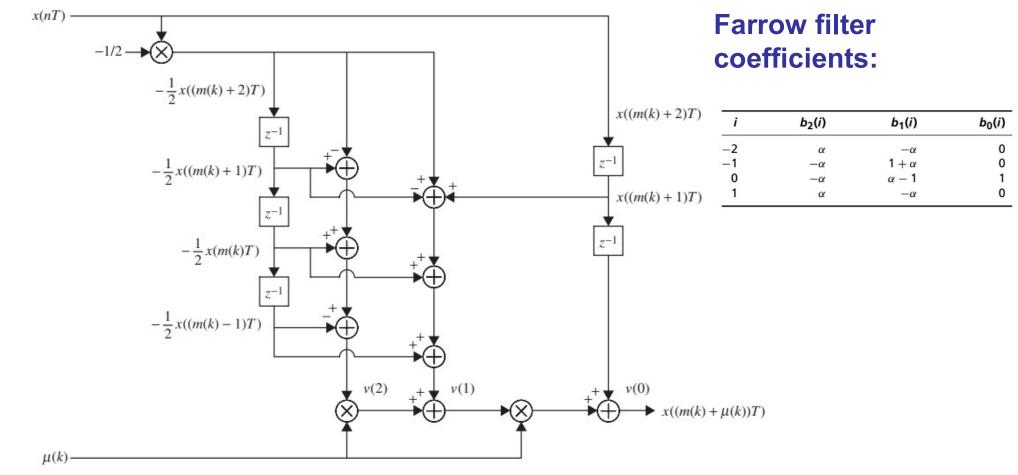
$$\begin{cases} h_{I}[-2+\mu_{k}] = \alpha \mu_{k}^{2} - \alpha \mu_{k} \\ h_{I}[-1+\mu_{k}] = -\alpha \mu_{k}^{2} + (\alpha+1)\mu_{k} \\ h_{I}[\mu_{k}] = -\alpha \mu_{k}^{2} + (\alpha-1)\mu_{k} + 1 \\ h_{I}[1+\mu_{k}] = \alpha \mu_{k}^{2} - \alpha \mu_{k} \end{cases}$$

• The interpolation equation becomes:

$$y(\mu_k) = \sum_{i=-2}^{1} x[(-i)]h_I[(i+\mu_k)]$$

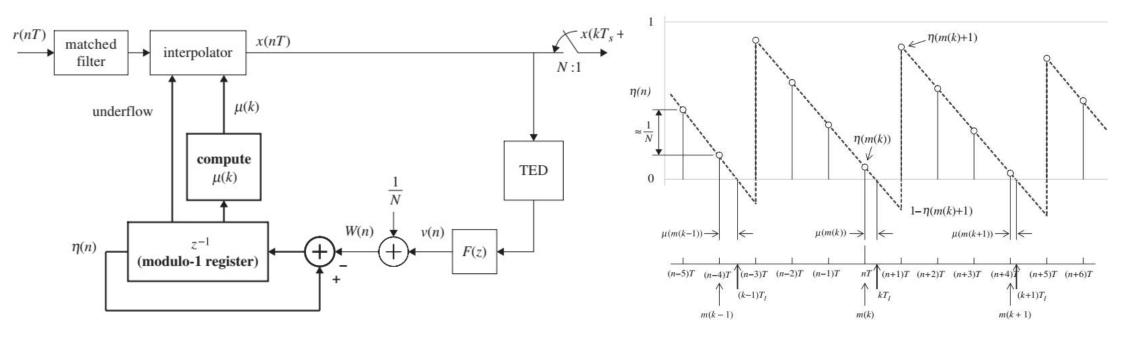
= $x[2](\alpha\mu_k^2 - \alpha\mu_k) + x[1](-\alpha\mu_k^2 + (\alpha+1)\mu_k)$
 $+x[0](-\alpha\mu_k^2 + (\alpha-1)\mu_k + 1) + x[-1](\alpha\mu_k^2 - \alpha\mu_k)$

□ The timing synchronizer: Interpolator structures



□ The timing synchronizer: Interpolator structures

• The Farrow interpolator is controlled by a modulo 1 counter:



Outline

- 1 2 – 3 –
- 4 GNU Radio demo with real hardware
- 5 -

QPSK system with the Adalm-Pluto

	×	Properties: Symbol Sync				
Gener	-	ocumentation	Advanced [General		
Timin		digital_symbol_sync_xx_0	ID			
TED SI		Complex -	О Туре	1/0		
Sam		Mueller and Müller	Timing Error Detector			
Expe		Modified Mueller and Mül	r Constellation	TED Slice		
3 		Zero Crossing	Samples per Symbol			
Loc		Gardner	Gan Expected TED Gain			
<u>Da</u> Maxi	Loop Bandwidth					
Output	ng Factor Mengali and D'Andrea GMSK					
Interpo		Maximum Deviation				
Filt	bo sgn(y[n])y'[n] Maximum Likelyhood					
F		MMSE, 8 tap FIR	ing Resampler	Interpolat		
-	Apply	OK				

		Prope	rties: Symbol Syr	nc	×		
General	Advanced	Docume	ntation				
Timing E	rror Detecto	Muell	er and Müller		•		
TED Slice	r Constellatio	n digital	l.constellation_qp	sk().base()			
Sample	s per Symbol	sps					
Expect	ed TED Gain	1.0					
Loop Bandwidth		width omega_n_norm					
Damp	Damping Factor zeta						
Maximu	Im Deviation	MMSE	, 8 tap FIR				
Output Sa	amples/Symb	ol Polyph	nase Filterbank, M	MSE			
Interpola	ting Resampl	er Polyph	Polyphase Filterbank, MF				
Filter	bank Arms	32	32				
PFB	MF Taps	۵					
			OK	Cancel	Apply		

Outline

2 -3 -4 -5 - Conclusion

1 -

5. Conclusion

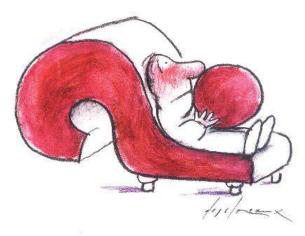
- Going from simulation to implementation adds (a lot of) complexity.
- Digital synchronization functions understanding and implementation require a good level in digital signal processing.
- □ Practical implementations are requested by today's students.
- □ For this particular purpose, GNU Radio is the right tool!

[1] M. Rice, "Digital Communications: a discrete time approach", 2nd Edition, Author, 2020.
[2] T. P. Zielinski, "Starting Digital Signal Processing in Telecommunication Engineering", Springer, 2021.

[3] <u>https://igorfreire.com.br/2016/10/15/symbol-timing-synchronization-tutorial/</u>

[4] <u>http://ricesimulink.groups.et.byu.net/</u>

Thank you for your attention!



Questions?