

GNSS Signal Processing Lab : GPS Receiver Case Study

Lorenzo Ortega

2022

1 GPS Receiver Position Computation

1.1 Trilateration : theoretical principle

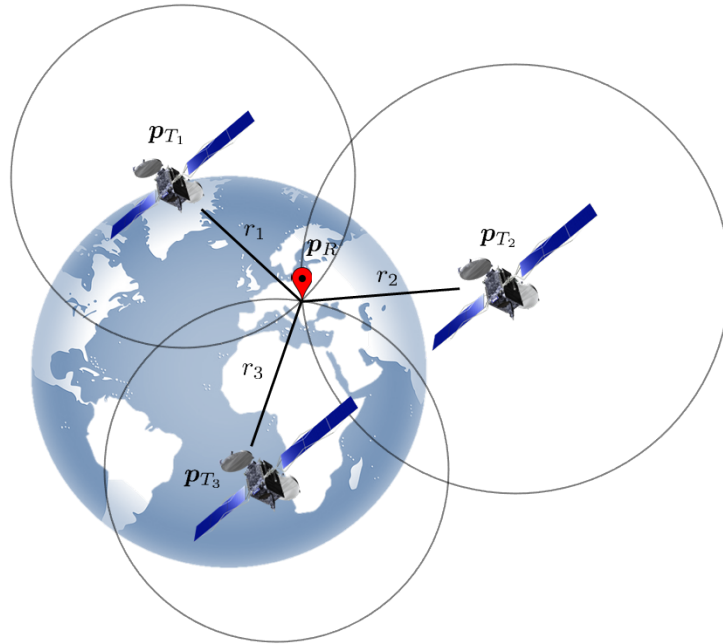


FIGURE 1 – GNSS positioning working principle : trilateration

Positioning via a constellation of satellites is the implementation of a known geometric property since antiquity, namely that the intersection of 3 spheres defines two points in the 3-dimensional space. This is the principle of trilateration : 3 satellites $\{T_1, T_2, T_3\}$ at known positions $\{\mathbf{p}_{T_1}, \mathbf{p}_{T_2}, \mathbf{p}_{T_3}\}$ define the center of 3 spheres, and the position \mathbf{p}_R of an observer on Earth is one of the two intersection points of these 3 spheres (the other one on space), which results in the following system of 3 (nonlinear) equations with 3 unknowns (\mathbf{p}_R) :

$$\begin{cases} r_1 = \|\mathbf{p}_{T_1} - \mathbf{p}_R\| \\ r_2 = \|\mathbf{p}_{T_2} - \mathbf{p}_R\| \\ r_3 = \|\mathbf{p}_{T_3} - \mathbf{p}_R\| \end{cases}, \quad (1)$$

which can be solved if the 3 distances $\{r_1, r_2, r_3\}$ are also known (see Figure 1). Then, we must be able to obtain the 3 distances $\{r_1, r_2, r_3\}$, which translates to measure the 3 travelling times

Thanks to Eric Chaumette & Jordi Vilà-Valls from ISAE SUPAERO

$\{\tau_1, \tau_2, \tau_3\} = \{\frac{r_1}{c}, \frac{r_2}{c}, \frac{r_3}{c}\}$ of an electromagnetic (EM) signal between each satellite and the observer that has a receiver adapted to perform this task (c is the speed of the EM waves = speed of light in the void).

Indeed, the propagation of an EM signal can be modeled (at first order) as the combination of an amplitude attenuation β (propagation budget) and a propagation time delay $\tau = \frac{r}{c}$, which leads to the following received signal model,

$$x(t) = \beta e(t - \tau) = \beta e\left(t - \frac{r}{c}\right), \quad |\beta| \ll 1, \quad \tau, r > 0. \quad (2)$$

The measurement of the propagation time between a transmitter T and a receiver R is obtained by calculating the cross-correlation function between the received signal $x(t)$ and the transmitted signal $e(t)$,

$$\widetilde{R}_{x,e}(\tau') = \int_{-\infty}^{+\infty} x(t) e(t - \tau')^* dt. \quad (3)$$

Question 1

Show that the maximum of $\left|\widetilde{R}_{x,e}(\tau')\right|$ is obtained for $\tau' = \tau$.

We therefore deduce that the propagation delay is identified by

$$\tau = \arg \max_{\tau'} \left\{ \left| \widetilde{R}_{x,e}(\tau') \right| \right\}. \quad (4)$$

But from (1), we need to measure 3 traveling times $\{\tau_1, \tau_2, \tau_3\} = \{\frac{r_1}{c}, \frac{r_2}{c}, \frac{r_3}{c}\}$. This is made possible by the simultaneous reception of *quasi-orthogonal* signals transmitted by the 3 satellites $\{T_1, T_2, T_3\}$,

$$x(t) = \beta_1 e_1(t - \tau_1) + \beta_2 e_2(t - \tau_2) + \beta_3 e_3(t - \tau_3) = \sum_{k=1}^3 \beta_k e_k(t - \tau_k), \quad (5)$$

which verify

$$\begin{aligned} \widetilde{R}_{x,e_1}(\tau') &= \int_{-\infty}^{+\infty} x(t) e_1(t - \tau')^* dt = \begin{cases} \beta_1 \int_{-\infty}^{+\infty} e_1(t - \tau_1) e_1(t - \tau')^* dt + \\ \beta_2 \int_{-\infty}^{+\infty} e_2(t - \tau_2) e_1(t - \tau')^* dt + \\ \beta_3 \int_{-\infty}^{+\infty} e_3(t - \tau_3) e_1(t - \tau')^* dt \end{cases} \\ \widetilde{R}_{x,e_1}(\tau') &\simeq \beta_1 \int_{-\infty}^{+\infty} e_1(t - \tau_1) e_1(t - \tau')^* dt, \end{aligned}$$

as well as

$$\widetilde{R}_{x,e_2}(\tau') \simeq \beta_2 \int_{-\infty}^{+\infty} e_2(t - \tau_2) e_2(t - \tau')^* dt, \quad \widetilde{R}_{x,e_3}(\tau') \simeq \beta_3 \int_{-\infty}^{+\infty} e_3(t - \tau_3) e_3(t - \tau')^* dt,$$

which makes possible to obtain 3 decoupled (“independent”) searches of the 3 propagation times. For each transmitter, the function

$$\begin{aligned}
 \int_{-\infty}^{+\infty} e_k(t - \tau_k) e_k(t - \tau')^* dt &= \int_{-\infty}^{+\infty} e_k(t - \tau_k) e_k(t - \tau_k - (\tau' - \tau_k))^* dt \\
 &\downarrow u = t - \tau_k, du = dt \\
 &= \int_{-\infty}^{+\infty} e_k(u) e_k(u - (\tau' - \tau_k))^* du = \int_{-\infty}^{+\infty} e_k(t) e_k(t - (\tau' - \tau_k))^* dt,
 \end{aligned}$$

only depends on the delay difference $(\tau' - \tau_k)$, that is,

$$\int_{-\infty}^{+\infty} e_k(t - \tau_k) e_k(t - \tau')^* dt = \widetilde{R}_{e_k}(\tau) \Big|_{\tau=\tau'-\tau_k}, \quad \widetilde{R}_{e_k}(\tau) = \int_{-\infty}^{+\infty} e_k(t) e_k(t - \tau)^* dt, \quad (6)$$

where $\widetilde{R}_{e_k}(\tau)$ is the so-called autocorrelation function of the signal $e_k(t)$.

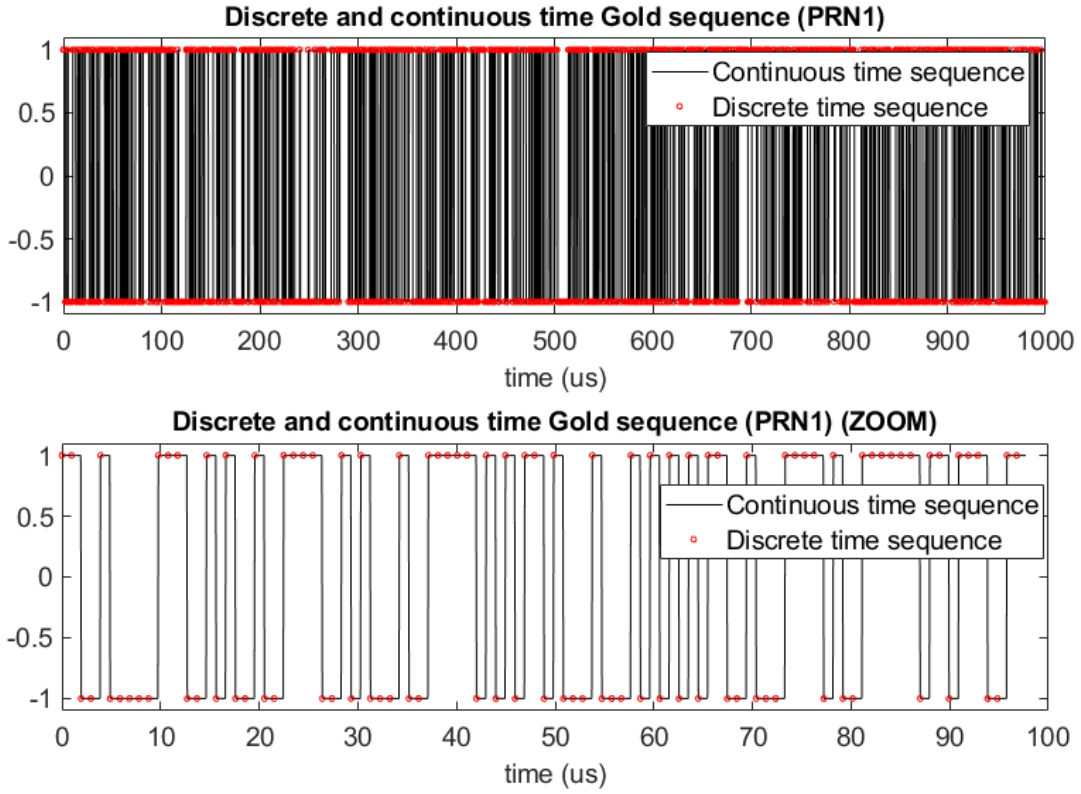


FIGURE 2 – Example of the digital-to-analog conversion (DAC) for the Gold code #1

A possible solution, being the one adopted in the legacy GPS L1 C/A signal, is to use Gold sequences (binary sequences) $\{c_k(n)\}_{n \in \mathbb{Z}}$ (see Figures 1-3 Matlab) converted into a continuous signal by a digital-to-analog converter (DAC),

$$\{c_k(n)\}_{n \in \mathbb{Z}} \xrightarrow{\text{DAC}} e_k(t) = \sum_{n \in \mathbb{Z}} c_k(n) \pi_{T_c}(t - (n-1)T_c), \quad \pi_{T_c}(t) = \begin{cases} 1 & \text{if } 0 \leq t \leq T_c \\ 0 & \text{otherwise} \end{cases} \quad (7)$$

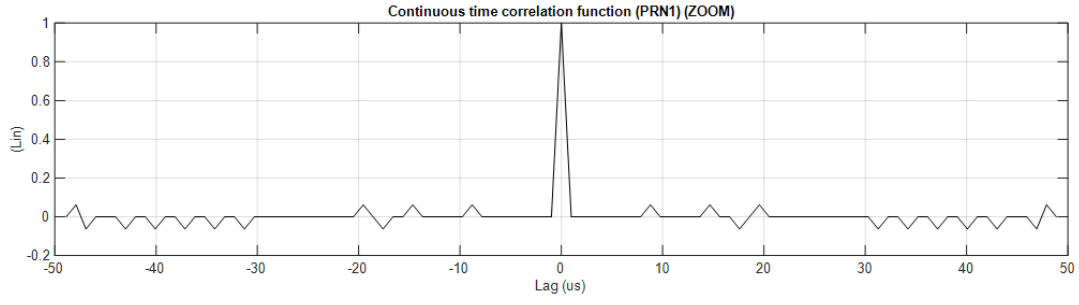
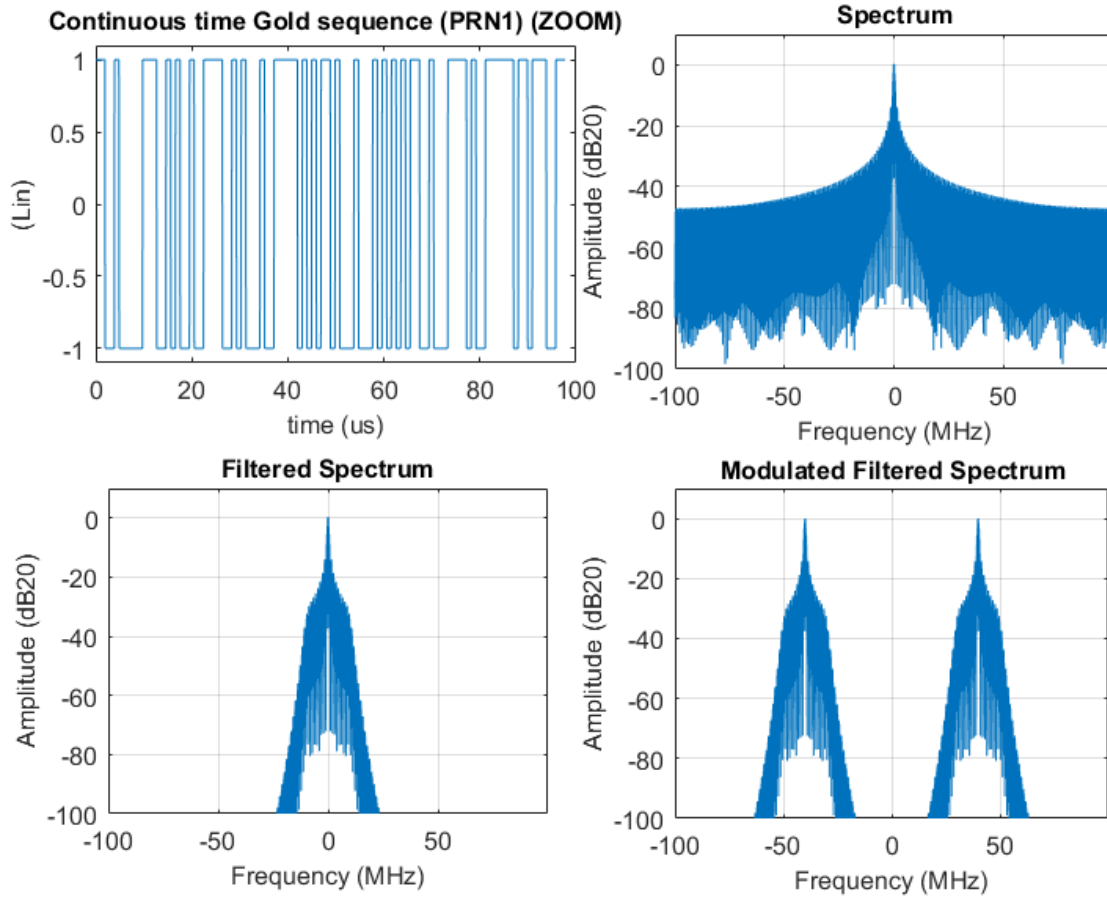


FIGURE 3 – Autocorrelation function $\widetilde{R}_{e_k}(\tau)$ (6) for $e_k(t) = e_1(t)$, as a function of τ (Lag).

1.2 Trilateration : practical implementation

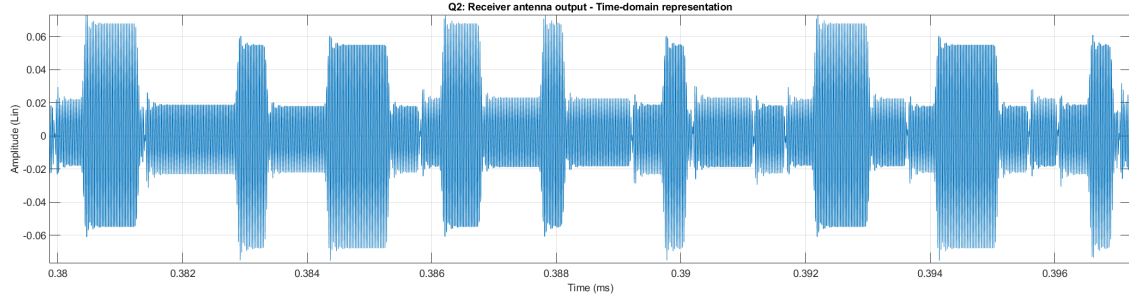
The propagation of an EM signal with “minimal” attenuation requires the modulation by a carrier frequency which makes it possible to transpose the spectrum of the signal $x(t)$, known as baseband signal (without carrier), to higher frequencies where the attenuation of the atmosphere (and other EM characteristics) is more favorable. The transmission electronics also contain a “shaping” filter (with impulse response $h_T(t)$) which makes it possible to limit the bandwidth (frequency support) of the transmitted signal in order to comply with international standards for sharing the frequency spectrum :

$$e_k(t) \xrightarrow[\text{filter } h_T(t)]{} \tilde{e}_k(t) = e_k(t) * h_T(t) \simeq e_k(t) \xrightarrow[\text{modulation } f_c]{} \tilde{e}_k(t) \cos(2\pi f_c t). \quad (8)$$



Finally, the signal at the reception antenna is written as (see Figure 4 Matlab) :

$$x_A(t) = \sum_{k=1}^3 \beta_k \tilde{e}_k(t - \tau_k) \cos(2\pi f_c(t - \tau_k)) = \sum_{k=1}^3 \beta_k \tilde{e}_k(t - \tau_k) \operatorname{Re} \{ e^{-j2\pi f_c \tau_k} e^{j2\pi f_c t} \}. \quad (9)$$



2 Radio frequency (RF) front-end architecture

From a real band-limited signal (BLS) of bandwidth B modeling the receiver antenna output (9) (see Figure (4) and Figure 4 Matlab) and an ADC operating at F_e (sampling frequency), **we are trying to reconstruct the baseband signal** (5) :

$$x(t) = \sum_{k=1}^3 \beta'_k \tilde{e}_k(t - \tau_k) \simeq \sum_{k=1}^3 \beta'_k e_k(t - \tau_k), \quad (10)$$

in order to be able to determine the propagation delays allowing to solve (1).

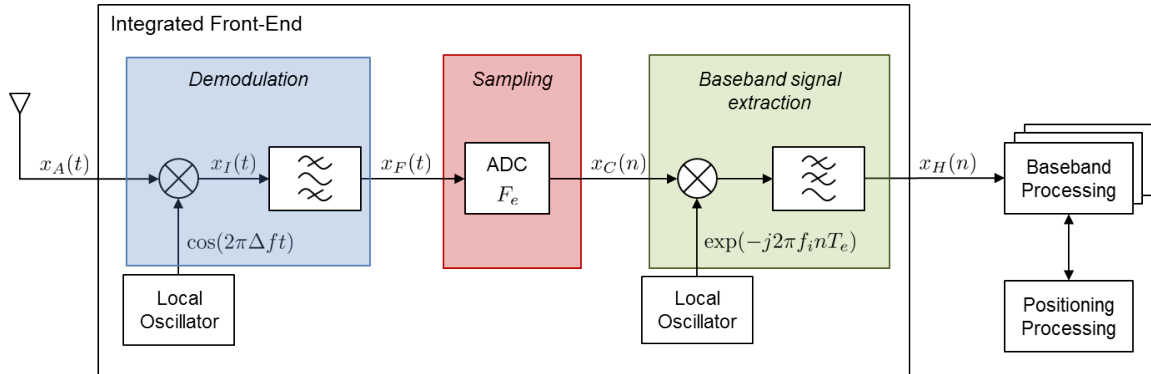


FIGURE 4 – GPS receiver RF front-end architecture.

Signal processing concepts

Nyquist-Shannon Theorem : real band-limited signal representation

If $x(t)$ is a real band-limited signal (BLS) in $[-\frac{B}{2}, \frac{B}{2}]$, then $\forall F_s \in \mathbb{R} \mid F_s > B$:

$$\begin{aligned} X(f) &= \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt = \sum_{n \in \mathbb{Z}} x(nT_s) e^{-j2\pi f n T_s T_s} \\ &\quad \downarrow \quad \uparrow \\ x(t) &= \sum_{n \in \mathbb{Z}} x(nT_s) \sin c(\pi F_s (t - nT_s)). \end{aligned}$$

What you must remember...

Any BLS $x(t)$ with bandwidth B can be decomposed on the **Shannon's orthogonal basis** (not orthonormal) (Shannon o.b.) $\{g_n(t) = \sin c(\pi F_s (t - nT_s))\}_{n \in \mathbb{Z}}$ iff $T_s < \frac{1}{B}$. In this basis, the coordinates of $x(t)$ are $x(nT_s)$.

The Fourier transform (FT) of a BLS is equal to its approximation by the rectangle mid point rule if the integration domain discretization is carried out with a sufficiently small constant step ($T_s < \frac{1}{B}$).



If $x(t)$ is not a BLS, then the projection of $x(t)$ over the Shannon o.b. is only an approximation of $x(t)$, where the approximation arises because of spectral aliasing,

$$\begin{aligned} f \in \left[-\frac{F_s}{2}, \frac{F_s}{2}\right] : \sum_{n \in \mathbb{Z}} x(nT_s) e^{-j2\pi f n T_s T_s} &= X(f) + \sum_{n \in \mathbb{Z}^*} X(f + nF_s) \\ &\quad \downarrow \uparrow \\ \sum_{n \in \mathbb{Z}} x(nT_s) \sin c(\pi F_s (t - nT_s)) & \end{aligned}$$

2.1 Demodulation

We want to perform a demodulation, i.e., move the useful part of the spectrum from the carrier frequency f_c to an intermediate frequency $f_i < f_c$ ($f_i = 15$ Mhz). This demodulation is carried out in two stages : a first stage translates the spectrum to the intermediate frequency, which is followed by the second state aiming to filter unwanted frequencies.

Question 2

→ Figure 4 Matlab

- Based on the graphical representation of the amplitude of the signal $x_A(t)$ spectrum in decibels (dB), what is the value, in MHz, of the carrier frequency f_c ?
- What is the frequency F_s used for the Shannon basis $\{g_n(t) = \sin c(\pi F_s (t - nT_s))\}_{n \in \mathbb{Z}}$?
- Is the frequency F_s sufficient to represent the signals at the antenna output $x_A(t)$ (9)?

Question 3

We want to transpose the received signal to an intermediate frequency $f_i < f_c$ (→ Figure 5 Matlab).

- Mathematically describe the transformation to be performed on $x_A(t)$ in order to obtain the transposed signal $x_I(t)$.
- Give the mathematical expression of $X_I(f)$, the spectrum of $x_I(t)$, as a function of $X_A(f)$, the spectrum of $x_A(t)$.

Question 4

We want to filter the high frequencies of the signal $x_I(t)$, since the useful signal is at the low frequencies.

Signal processing concepts

Linear time-invariant (LTI) systems

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df \quad \xrightarrow{\boxed{L}} \quad \begin{aligned} y(t) &= \int_{-\infty}^{\infty} X(f) H(f) e^{j2\pi ft} df \\ y(t) &= \int_{-\infty}^{\infty} x(u) h(t-u) du = x(t) * h(t) \end{aligned}$$

where $H(f)$ is the “transfer function” (in the frequency domain), and its inverse FT $h(t) \Leftarrow H(f)$ is the “impulse response” (in the time domain).

If $x(t)$ is a **BLS** in $[-\frac{B}{2}, \frac{B}{2}]$, then $\forall F_s \in \mathbb{R} \mid F_s > B$:

$$\begin{aligned} (x * h)(t) &= \sum_{n \in \mathbb{Z}} (x * h)(nT_s) \sin c(\pi F_s(t - nT_s)) \\ (x * h)(nT_s) &= T_s \sum_{l \in \mathbb{Z}} x(lT_s) h((n-l)T_s) \end{aligned}$$

- Express $x_I(t)$ from the expression of $x_A(t)$ (9) in such a way as to reveal the low frequency components $x_I^-(t)$ and high frequency components $x_I^+(t)$ (\rightarrow Figure 5 Matlab) such as :

$$x_I(t) = x_I^-(t) + x_I^+(t)$$

- What is the filtering operation performed by a LTI system ?
- Express the signal $x_F(t)$ obtained at the output of the filter as a function of $x_I(t)$ and $h_I(t)$.
- Deduce the expression of the FT of $x_F(t)$.
- What properties should an analog filter verify ? Does the considered filter (\rightarrow Figure 6 Matlab) verify them ?
- By observing the spectrum of the filtered signal (\rightarrow Figure 7 Matlab), what can you say from a qualitative point of view about this demodulation operation ?
- If the demodulation operation was ideal, what would be $x_F(t)$?

What you must remember...



For any BLS $x(t)$ with bandwidth B , any transformation by a LTI system (filter,...) can be expressed from the transformation of its co-ordinates $\{x(nT_s)\}_{n \in \mathbb{Z}}$ in the **Shannon o.b.**

2.2 Analogique-to-Digital Conversion

We want to simulate the ADC operation (see Figure 4) with sampling frequency $F_e = 50$ MHz.

Signal processing concepts

Nyquist-Shannon Theorem : sampling real signals

Any sequence of time samples $\{x(nT_e)\}_{n \in \mathbb{Z}}$ obtained from a real signal $x(t) \Leftrightarrow X(f)$, is associated with a continuous real signal with band-limited bandwidth $F_e = \frac{1}{T_e}$:

$$x_e(t) = \sum_{n \in \mathbb{Z}} x(nT_e) \sin c(\pi F_e(t - nT_e)) \Leftrightarrow \begin{cases} X_e(f) = \sum_{n \in \mathbb{Z}} x(nT_e) e^{-j2\pi f n T_e} T_e \\ X_e(f) = \sum_{l \in \mathbb{Z}} X(f + lF_e) \end{cases}$$

What you must remember...



If the continuous signal $x(t)$ at the input of the ADC is of bandwidth B , and if the sampling frequency of the ADC verifies $F_e < B$, then there is spectral aliasing.

Question 5

- We note $x_C(n)$ the samples at the output of the ADC (\rightarrow Figure 8 Matlab). Express the sampled FT $X_C(f)$ associated to the samples $x_C(n)$ as a function of $X_F(f)$.
- What is the sampling effect on the spectrum of $X_F(f)$?
- If the demodulation had been ideal, what would be $X_C(f)$?
- What would a priori be the best choice for f_i ? In this case what would “ideally” be $x_C(t) \Leftrightarrow X_C(f)$?
- Are the frequencies f_i and F_e well suited one to the other?

Question 6

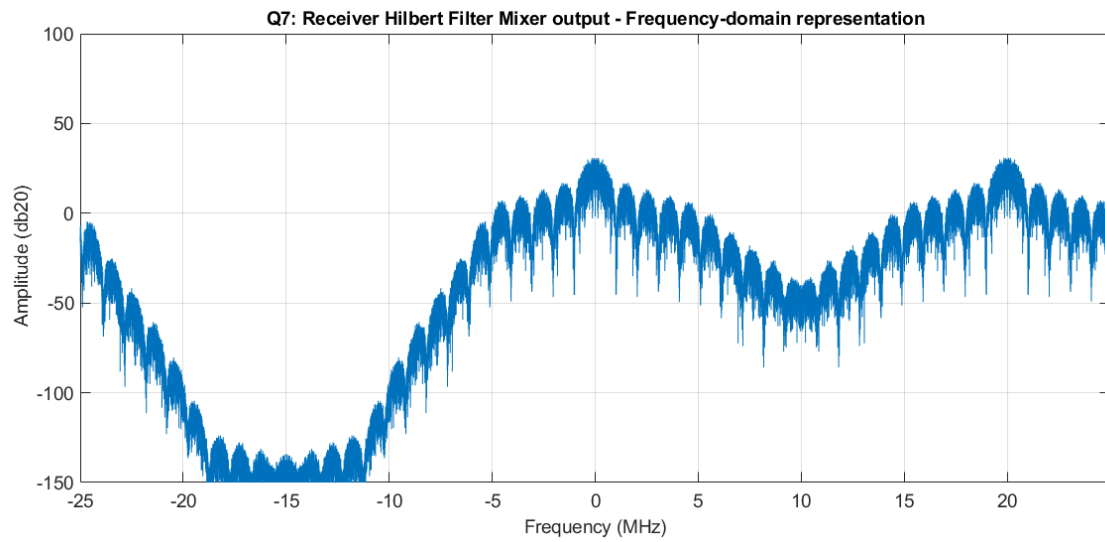
Show that there are two sequences $x_C^+(n)$ et $x_C^-(n)$ such that :

$$X_C(f) = \sum_{n \in \mathbb{Z}} x_C(n) e^{-j2\pi f n T_e} = X_C^+(f - f_i) + X_C^-(f + f_i), \quad \left| \begin{array}{l} X_C^+(f) = \sum_{n \in \mathbb{Z}} x_C^+(n) e^{-j2\pi f n T_e} T_e \\ X_C^-(f) = \sum_{n \in \mathbb{Z}} x_C^-(n) e^{-j2\pi f n T_e} T_e \end{array} \right.$$

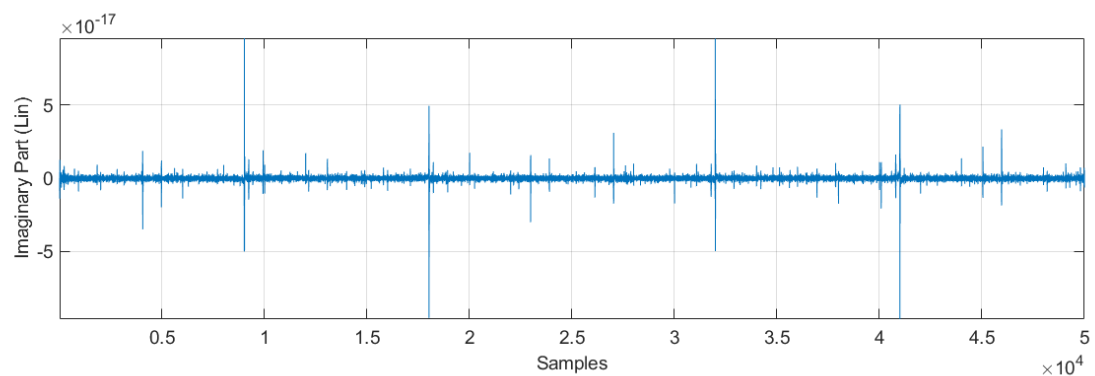
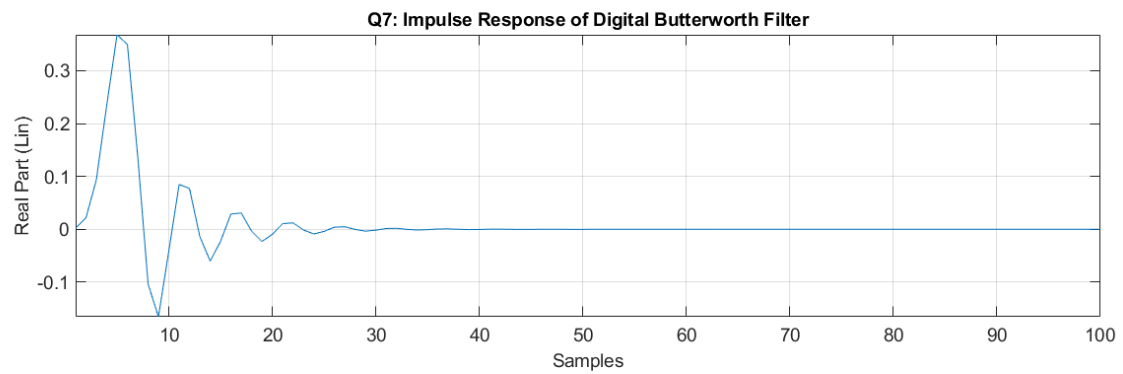
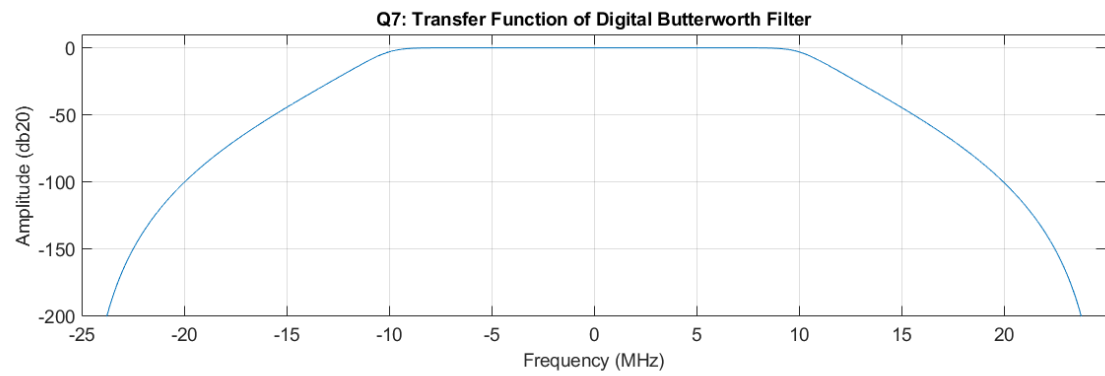
2.3 Digital Filtering

We want to preserve only $x_C^+(n)$. For this we use a digital Hilbert filter (\rightarrow Figure 10 Matlab) : frequency shift $-f_i$

$$\begin{aligned} x_C(n) &= x_C^+(n) e^{j2\pi f_i n T_e} + x_C^-(n) e^{j2\pi(-f_i) n T_e} \\ &\downarrow \\ x_C(n) e^{-j2\pi f_i n T_e} &= x_C^+(n) + x_C^-(n) e^{j2\pi(-2f_i) n T_e} \Leftrightarrow \begin{cases} X_C^+(f - f_i) \rightarrow X_C^+(f) \\ X_C^-(f + f_i) \rightarrow X_C^-(f + 2f_i) \end{cases} \end{aligned}$$



followed by a digital low-pass filter :



Generally speaking, the Hilbert filter is a LTI system which allows, from a real passband signal $x(t)$, to construct the complex signal $z(t)$ (so-called complex analytic signal) whose spectrum $Z(f)$ coincides with $X(f)$ for positive frequencies :

$$x(t) = \text{Re}\{z(t)\} = \frac{z(t) + z^*(t)}{2} \Leftrightarrow X(f) = \frac{Z(f)}{2} + \frac{Z(-f)^*}{2} / \forall f < 0 : Z(f) = 0. \quad (11)$$

Question 7

By observing the spectrum of the filtered signal (\rightarrow Figure 10 Matlab), what can you say qualitatively about this Hilbert filtering operation ?

3 Exploiting the baseband signal to determine the propagation delays

At the output of the Hilbert filter, ideally we obtained :

$$x_H(n) \simeq x_C^+(n) = \sum_{k=1}^3 \beta'_k \tilde{e}_k(nT_e - \tau_k) \simeq \sum_{k=1}^3 \beta'_k e_k(nT_e - \tau_k), \quad \beta'_k \in \mathbb{C} \quad (12)$$

$$e_k(n) = \sum_{l=0}^{1022} c_k(l) \pi(n - lM), \quad \pi(n) : \begin{cases} 1 & \text{if } 0 \leq n \leq M-1 \\ 0 & \text{otherwise} \end{cases}, \quad c_k(l) \in \{-1, 1\} \quad (13)$$

where $MT_e = T_c$ and T_c is the “chip” duration, that is, of a GPS code symbol. Each GPS code is a Gold sequence with 1023 symbols and a total duration equal to $10^{-3}s$. Then, $M = \frac{10^{-3}}{1023} F_e \simeq 49$ (see Figure 5).

Recall that the samples $\{x_H(n)\}_{n \in \mathbb{Z}}$ and $\{e_k(n)\}_{n \in \mathbb{Z}}$ are the coordinates of the BLS $x_H(t)$ and $e_k(t)$ on the Shannon's o.b. with bandwidth F_e :

$$x_H(t) = \sum_{n \in \mathbb{Z}} x_H(n) \sin c(\pi F_e(t - nT_e)), \quad e_k(t) = \sum_{n \in \mathbb{Z}} e_k(n) \sin c(\pi F_e(t - nT_e)) \quad (14)$$

Question 8

Compare the “measured” cross-correlation functions (obtained from the measured $x_H(t)$ (14), \rightarrow Figures 201-203 Matlab)

$$\widetilde{R}_{x_H, e_k}(\tau') = \int_{-\infty}^{+\infty} x_H(t) e_k(t - \tau')^* dt$$

and the “ideal” autocorrelation functions (\rightarrow Figures 101-103 Matlab)

$$\widetilde{R}_{e_k}(\tau') = \int_{-\infty}^{+\infty} e_k(t) e_k(t - \tau')^* dt$$

Question 9

Do the results obtained for $\widetilde{R}_{x_H, e_1}(\tau), \dots, \widetilde{R}_{x_H, e_3}(\tau)$ validate the quasi-orthogonality hypothesis of the signals $e_k(t)$?

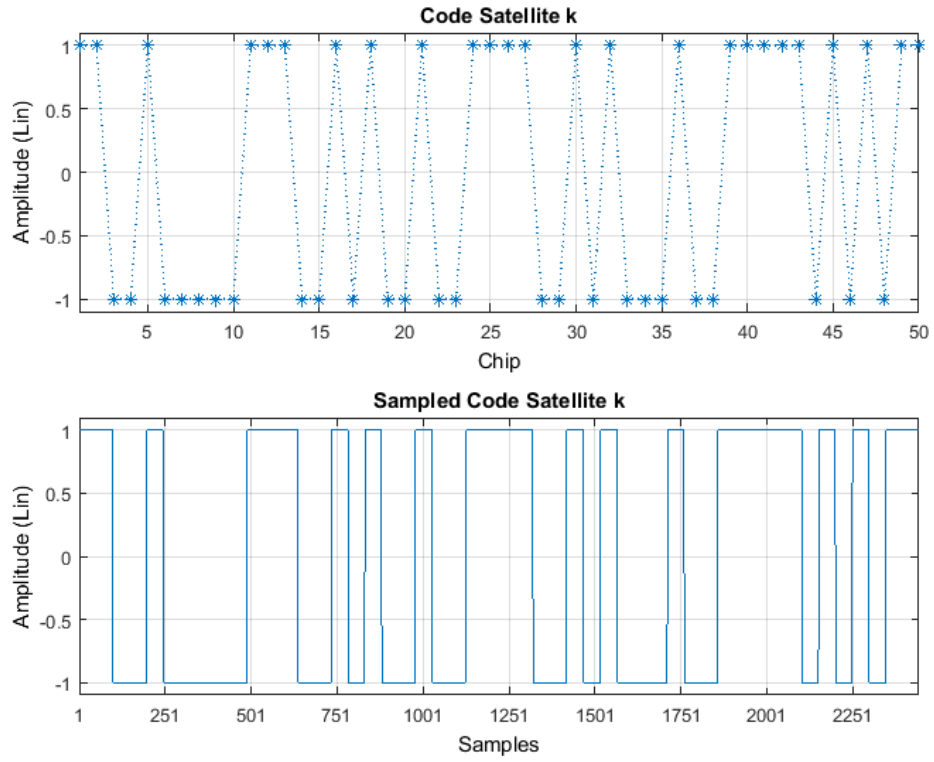


FIGURE 5 – GPS code $\{c_k(l)\}_{l \in [0, 1022]}$ for satellite k and the corresponding baseband signal sampled at $F_e : \{e_k(n)\}_{n \in [0, 1023 \times 49 - 1]}$.