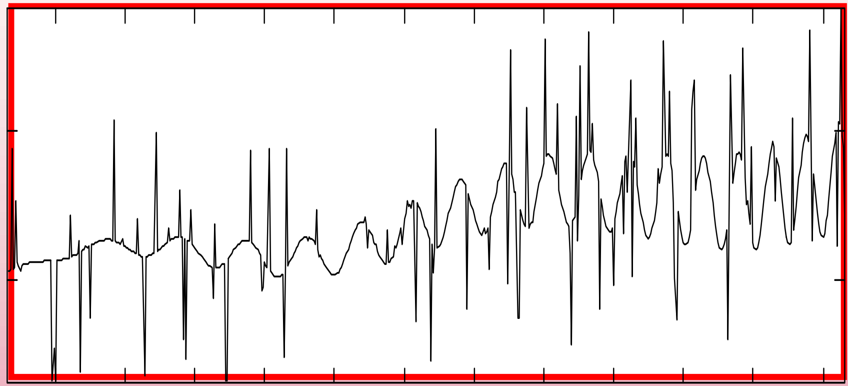
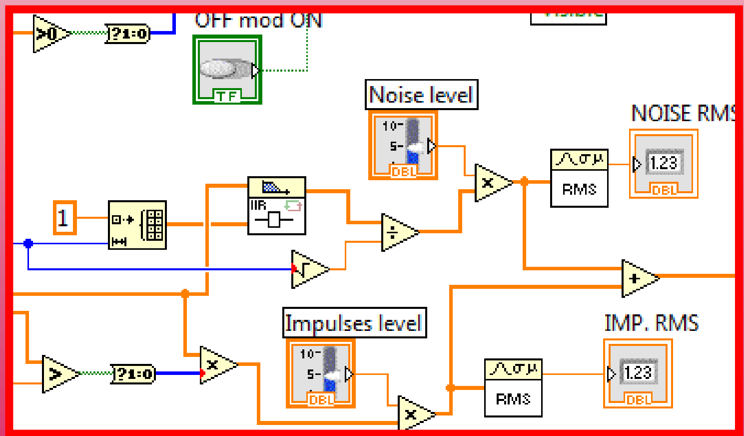
Analog and Digital Signal Processing





ORGANISATION DU COURS/LABORATOIRE

Analog and Digital Signal Processing

FORWORDS:

A critical examination of today's signal processing in the current literature and academic teaching prompts the following observations:

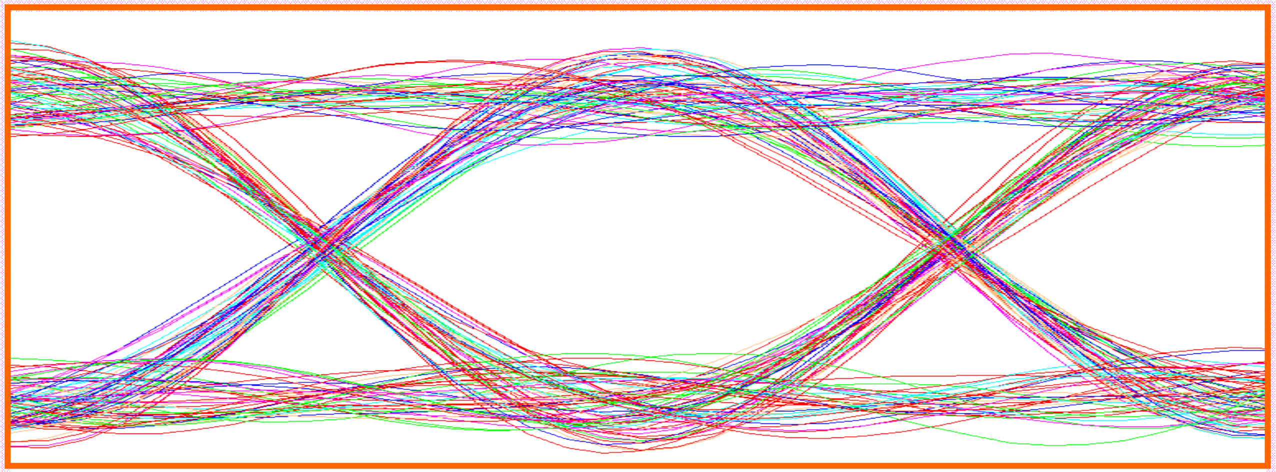
1. Computer simulations, when properly applied, provide a great deal of insight into a problem of interest, but they are **no substitute for tests with real-life data**. It is therefore not surprising that many algorithms fail to survive the ″test of time″.

2. Without question, mathematics is a powerful tool that gives an algorithm both elegance and general applicability. By the same token, however, an algorithm that **ignores physical reality** may end up being of **limited** or **no practical use**.

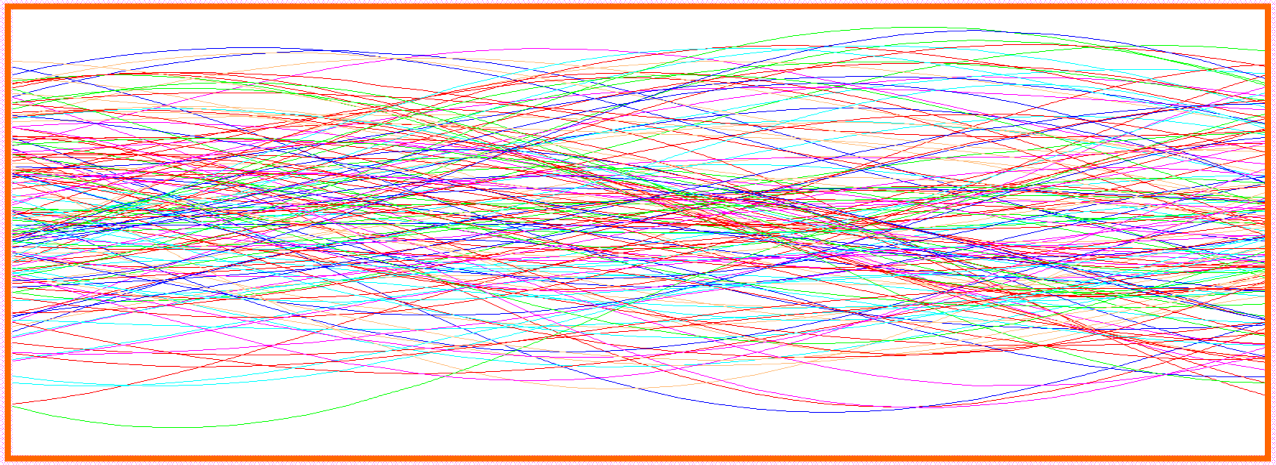
3. **Signal processing is at its best** when it successfully combines the unique ability of mathematics to generalize with both the insight and **prior information** gained from the underlying physics of the problem at hand.

IEEE Signal Processing Magazine
Simon Haykin, McMaster University
Hamilton, Ontario, Canada

Your brain before….



…. Your brain after



Introduction to Signal Processing

The general problem of Signal Processing is depicted with the following figure:

Signal Pre-Processing	Information source	Human voice Physical phenomena
	↓ Measurement or observation	Microphone Transducer Analog Preamplifier
Signal Processing Analog, Digital or Mixed Mode	↓ Signal representation	A→D converter
	↓ Signal transformation	Digital Filtering Spectral Analysis (FFT) Correlation……..
	↓ Extraction and utilization of information	Voice Recognition Frequency, Distance, Active Power Estimation

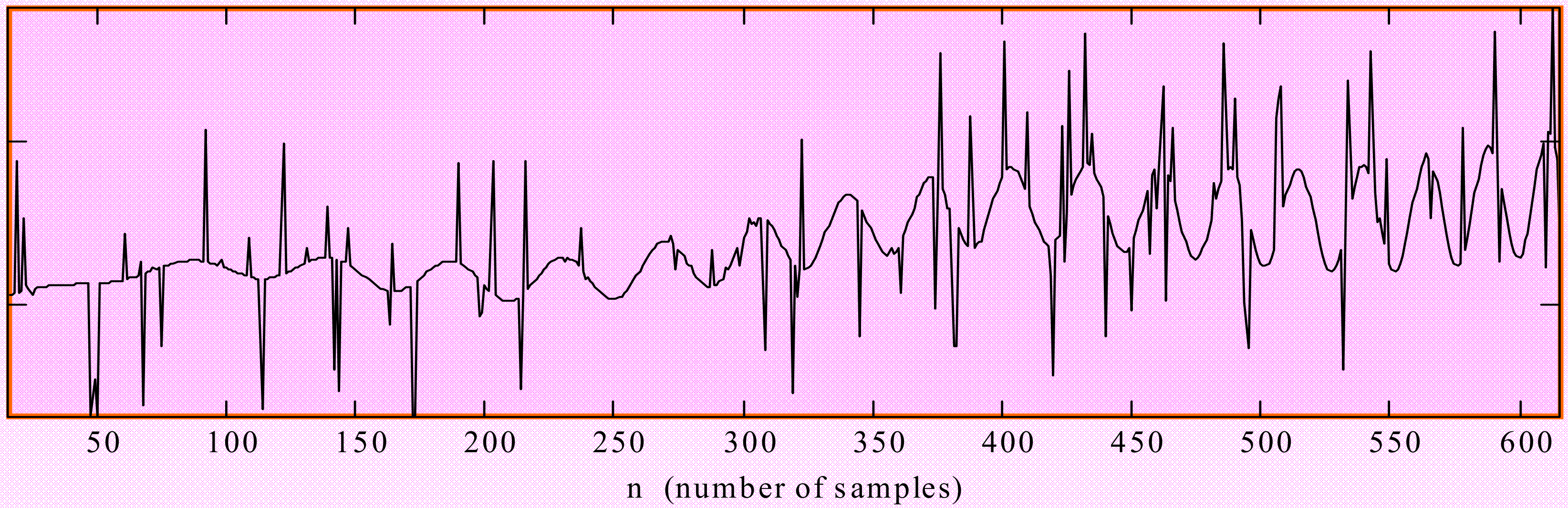
SIGNAL PROCESSING APPLICATIONS:

Instrument:	Spectrum analysis – Transient analysis
High speed control:	Robotics – Assembly line
Telecommunication:	GSM – CDMA – GPS – Blue-Tooth ……..
Physics, Medical:	Seismic warning – Scanner – Ultrasound
Military:	Electronic counter-measure - Missile
Image processing:	Fingerprint – MPEG – Pattern recognition
Speech processing:	Authentication – Compression
Consumer:	HDTV – CDs – DVDs – MP3 ……………
Automotive:	Anti skid – Engine control
Power:	Power plant – Grid supervision

CLASSIFICATION OF SOME SIGNAL PROCESSING GOALS

- *Estimation , Filtering*
- *Detection , Classification*
- *Coding , Encryption*
- *Modulation / Demodulation*
- *Synthesis, Compression*
- *Perceptual Enhancement*

ESTIMATION, FILTERING: *Example #1*



What characterizes this signal?

Sine-wave of increasing frequency

Low frequency sine-wave

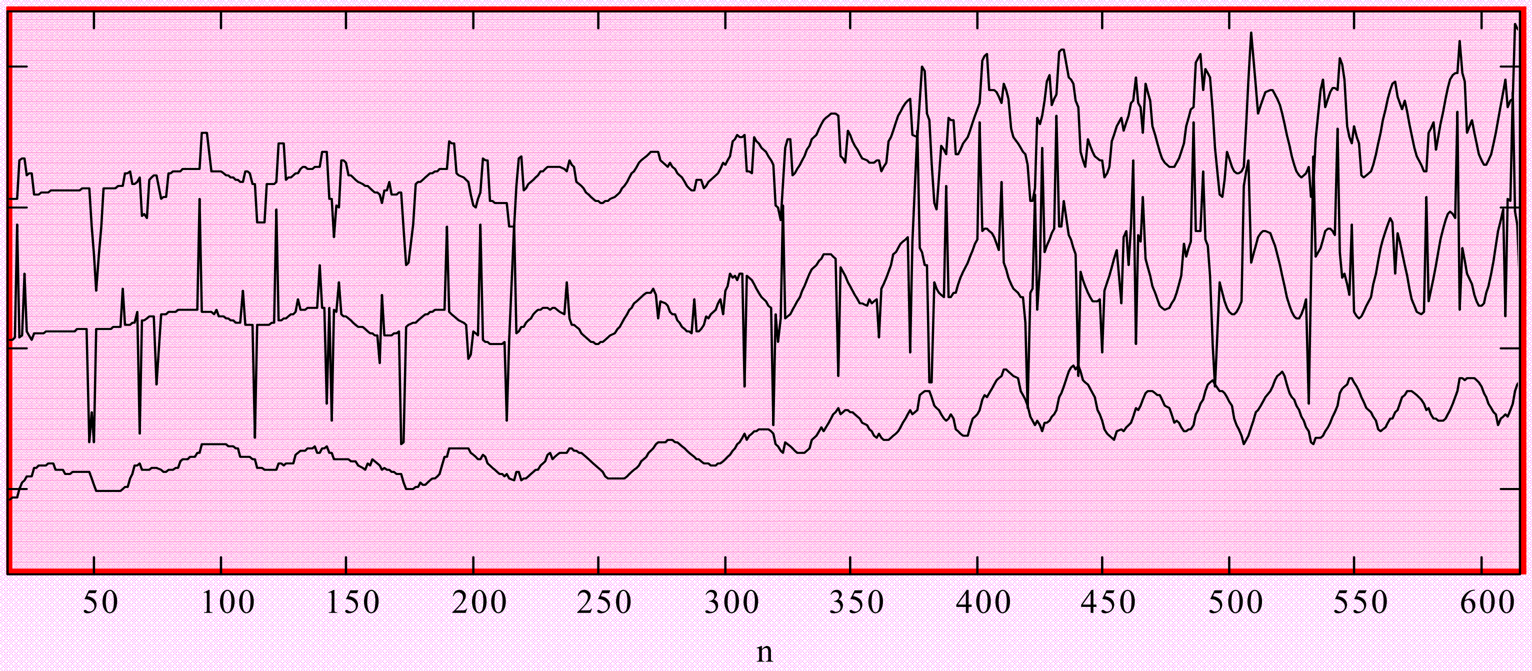
DC component

Impulsions

$$x(n) := 0.3 \cdot \left[\frac{n}{50} + 2 \cdot \sin(0.02 \cdot n) + 0.01 \cdot n \cdot \sin[0.08 \cdot n \cdot (1 + 0.002 \cdot n)] \right] + \mathrm{Imp}(n)$$

Running averager: $x_{run3}(n) = (1/3)\,[x(n) + x(n-1) + x(n-2)]$

Top: run3, center: x(n), bottom: run15

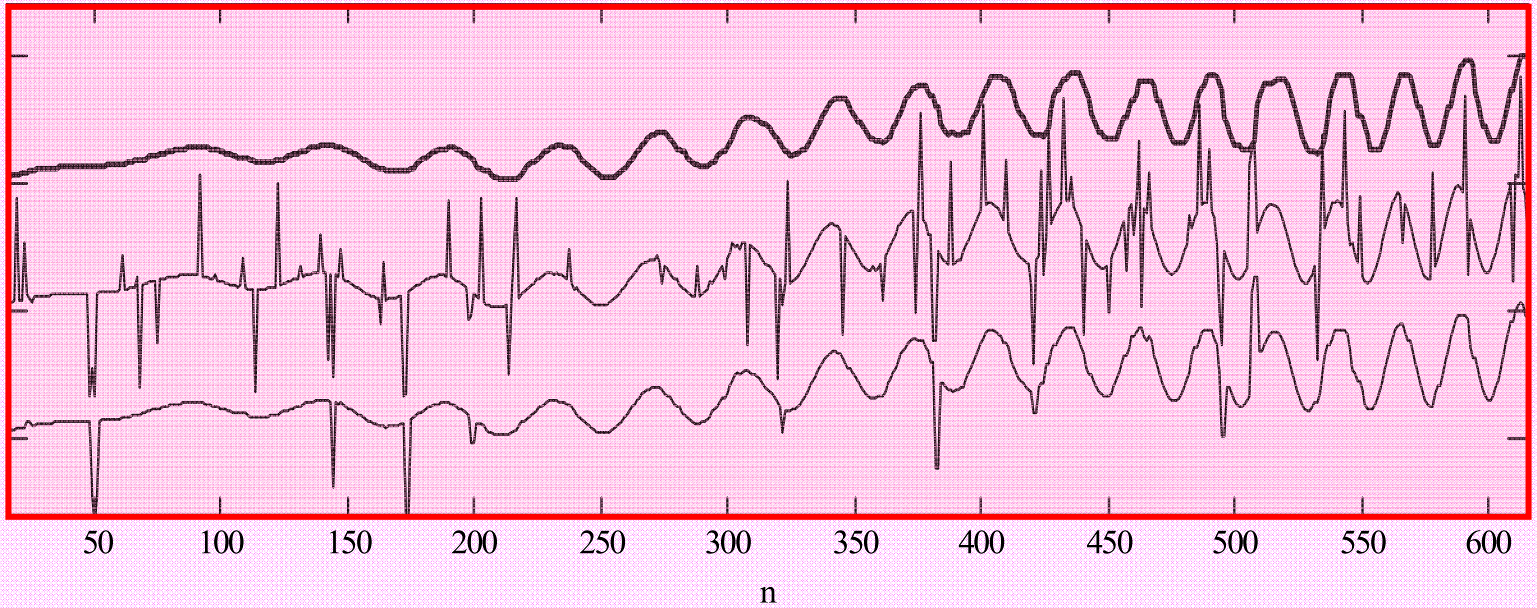


Comment: Not suited to impulsive noise filtering!

Median filter: $x_{med3}(n)$ = Median[x(n), x(n-1), x(n-2)]

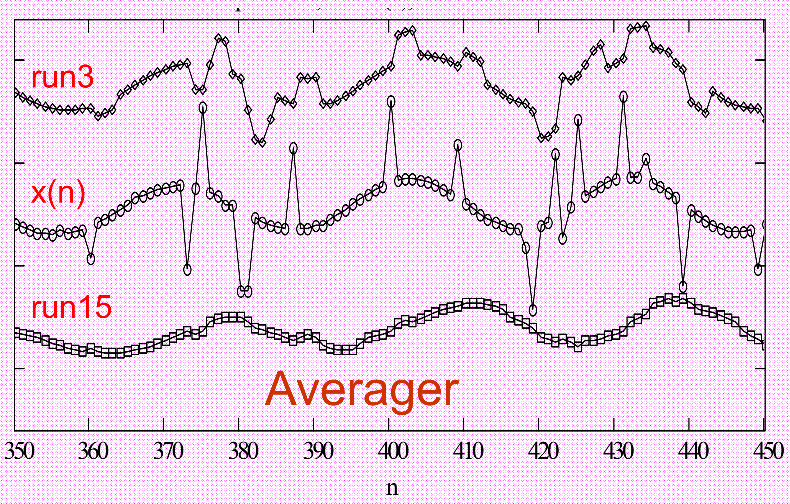
e.g. Median[2,9,8] = 8, Median[0,4,0] = 0

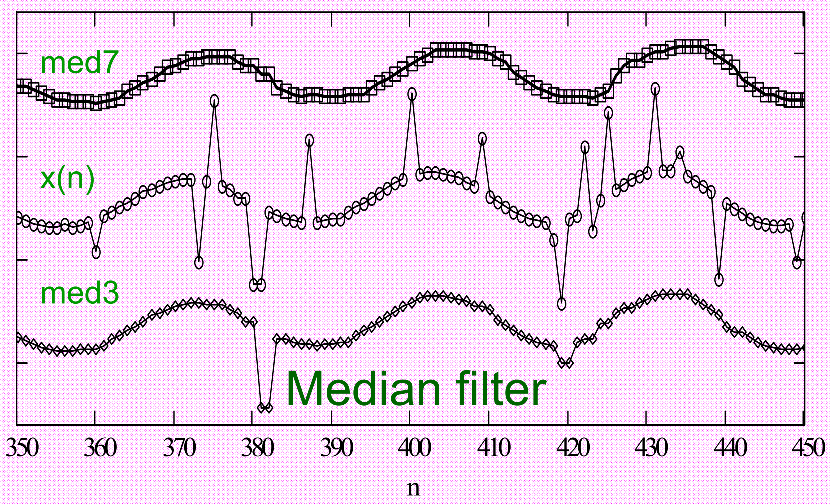
Top: med7, center: x(n), bottom: med3



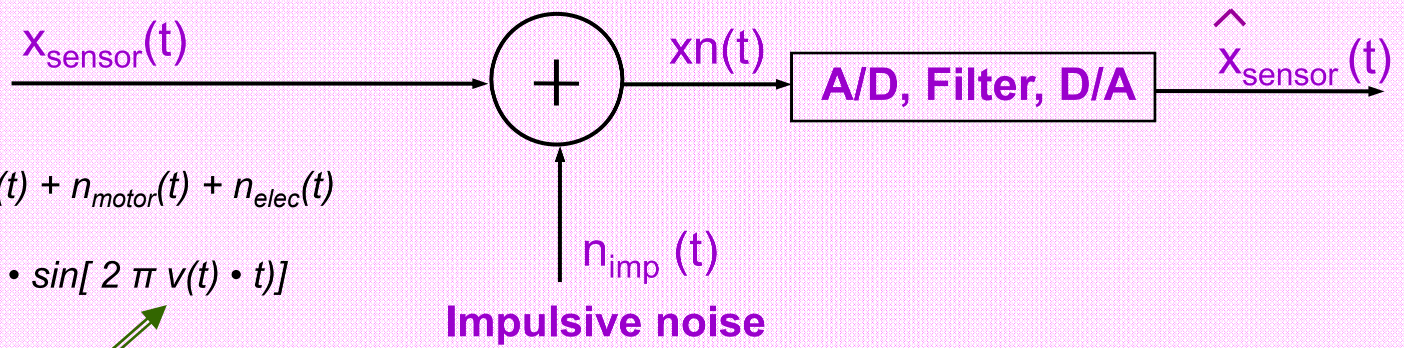
Median filter: Well suited to impulsive noise

A more detailed view:





Example #1 modelization: velocity sensor

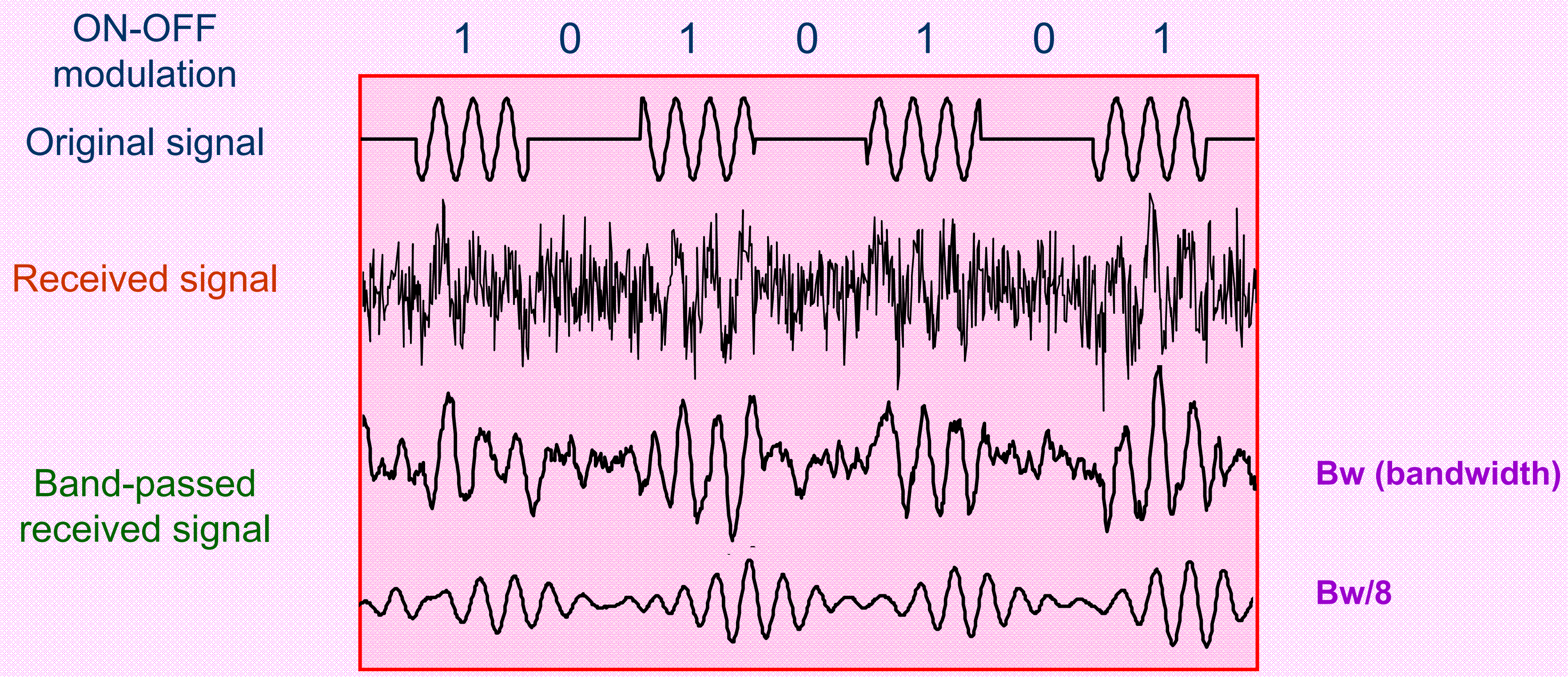


$x_{sensor}(t) = f(t) + n_{motor}(t) + n_{elec}(t)$

$f(t) = \alpha(t) \cdot \sin[\, 2\pi\, v(t) \cdot t)]$

Velocity

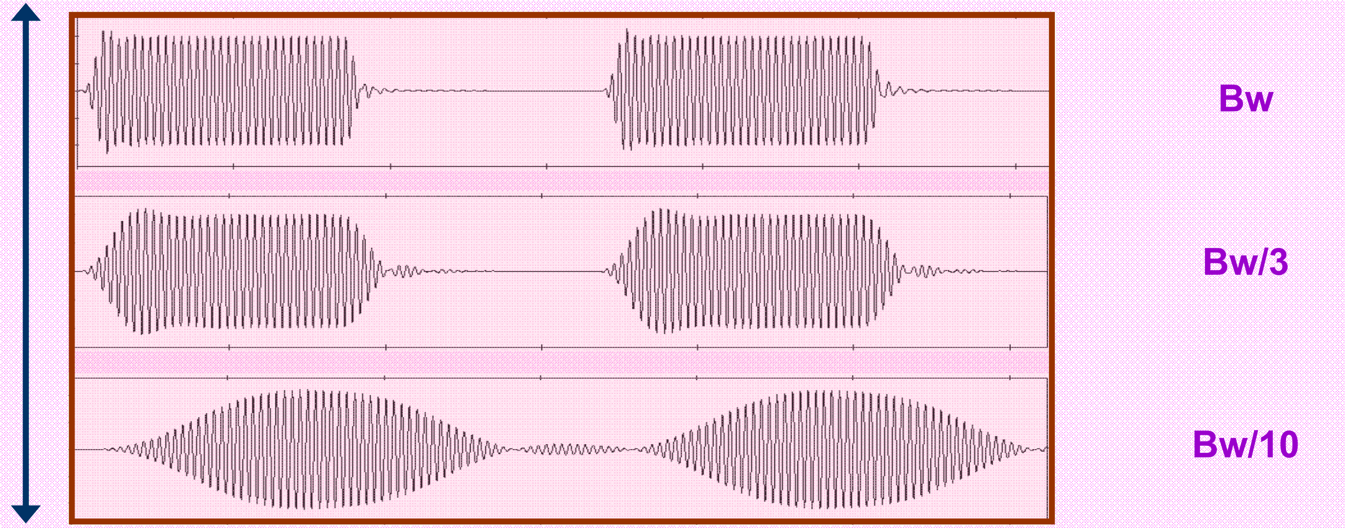
ESTIMATION, FILTERING: *Example #2*



Filtering consequences: Delay

Rise-time and fall-time increase

Band-passed signals



In summary:

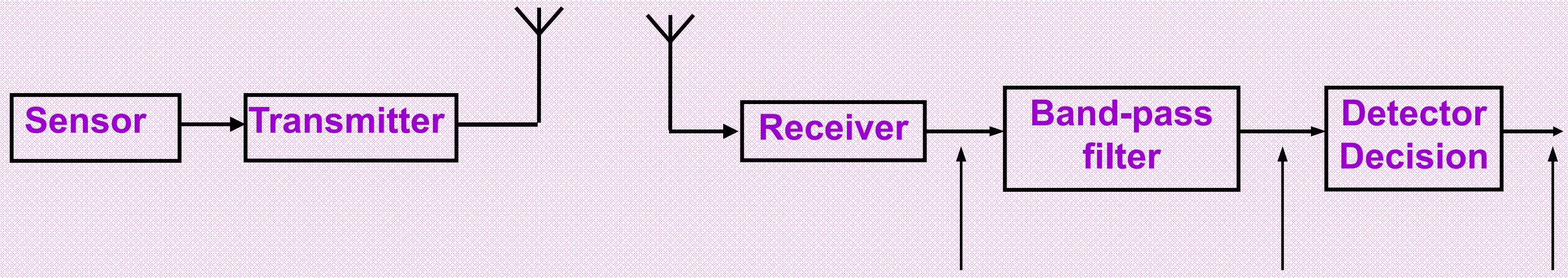
*The choice of the estimation and/or filtering strategy yielding the best result is obtained when as **many characteristics** as possible of both, the noise and the signal, are known and when "what we want to know of the desired signal" is clearly defined.*

DETECTION, CLASSIFICATION: *Example #1*

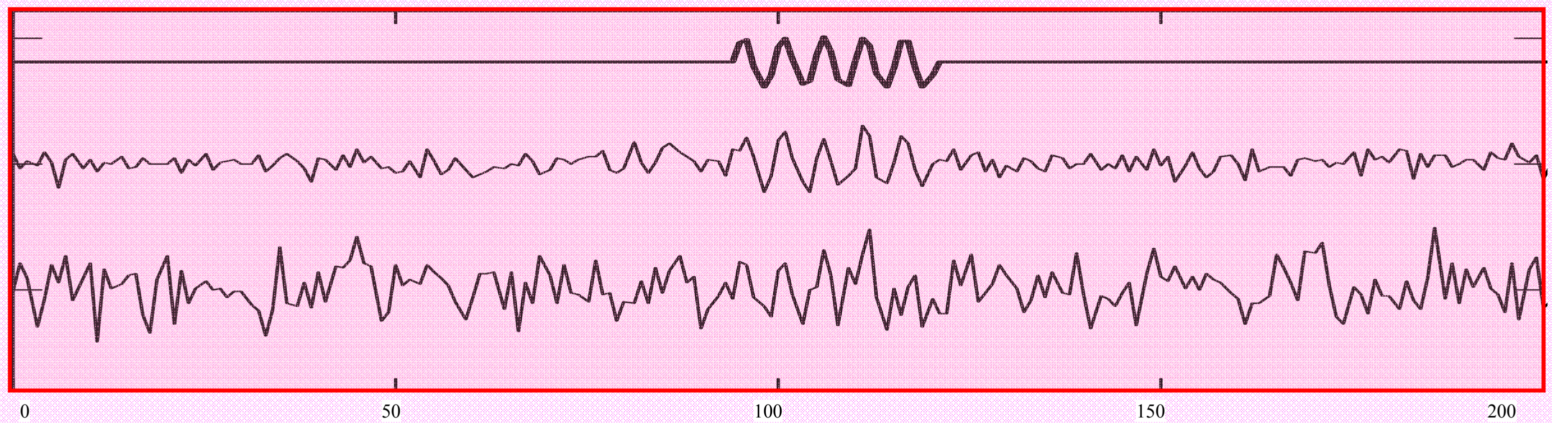
Detection

An alarm signal is characterized as follows:

One single 0.5 μs burst of a 10 MHz sine-wave (5 cycles)

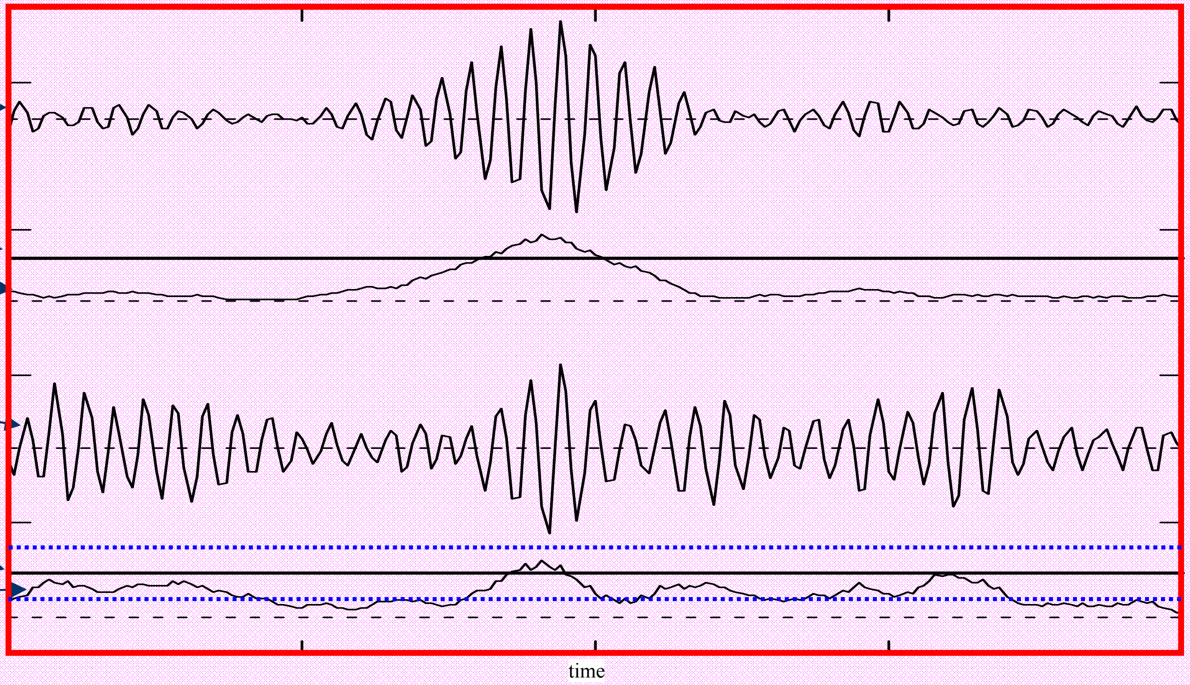


xr(t) top: No noise, middle: moderate noise, bottom: heavy noise



xbpf(t)
(moderate noise)
Threshold
Env[xbpf(t)]

xbpf(t)
(heavy noise)
Threshold
Env[xbpf(t)]



4 possible outcomes:

No alarm:	**No detection**
	Detection
Alarm transmitted	**No detection**
	Detection

FINALLY:

Errors due to threshold setting

Reliability improvement?

- **Increase the amplitude**
- **Increase the burst duration**
- **Repeat the burst**
- **Use frequency diversity**
- **……………………………**

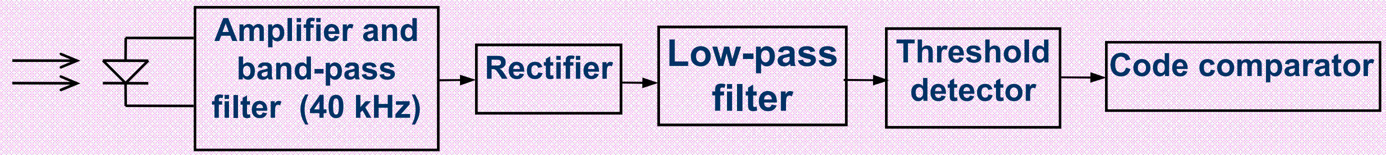
➔ Cost: MORE ENERGY

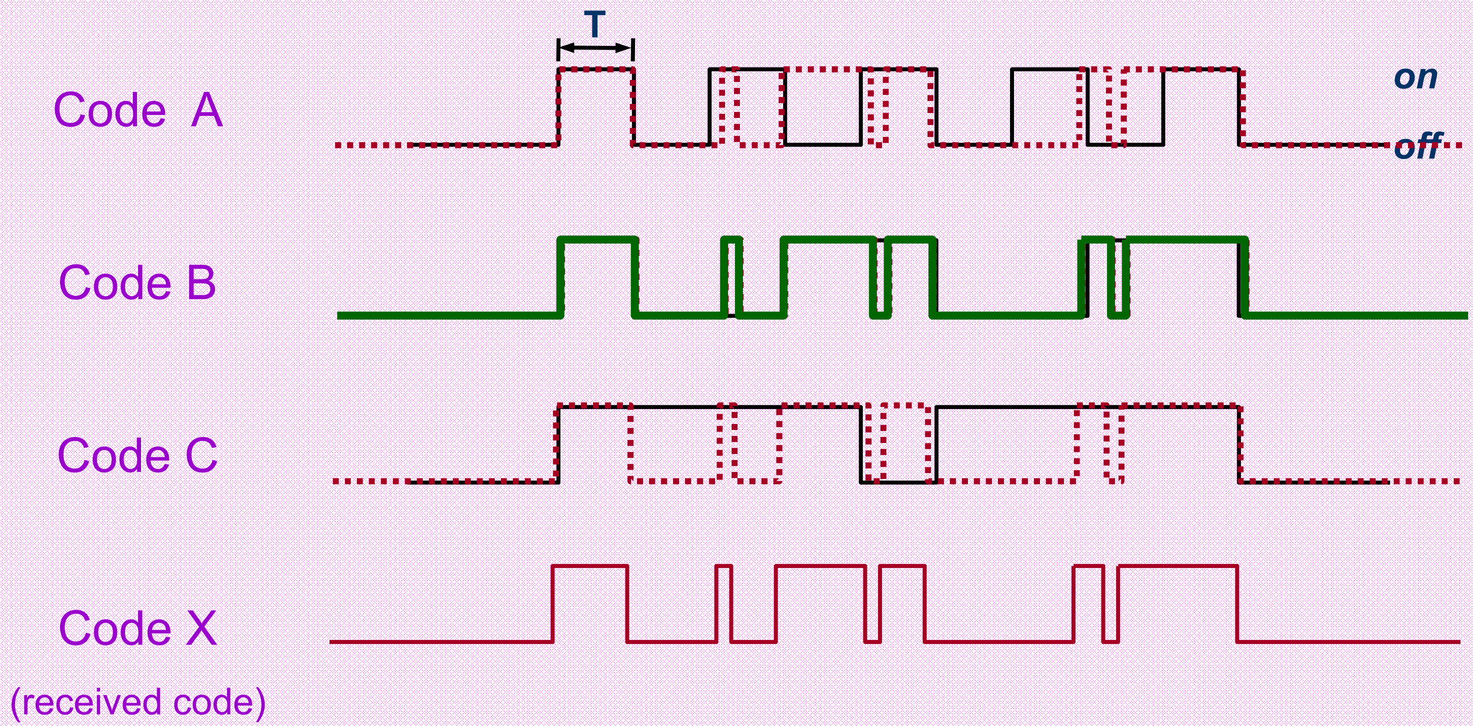
- **Select a « better » frequency**
- **Improve the transmitter antenna location and/or antenna gain**
- **Improve the receiver antenna location and/or antenna gain**
- **Adaptative threshold**
- **…………………………**

➔ Cost: HARDWARE + TIME

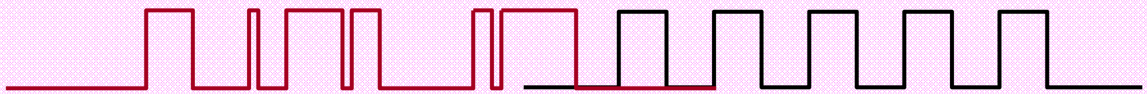
DETECTION, CLASSIFICATION: *Example #2*

Classification: Infra-red remote control receiver

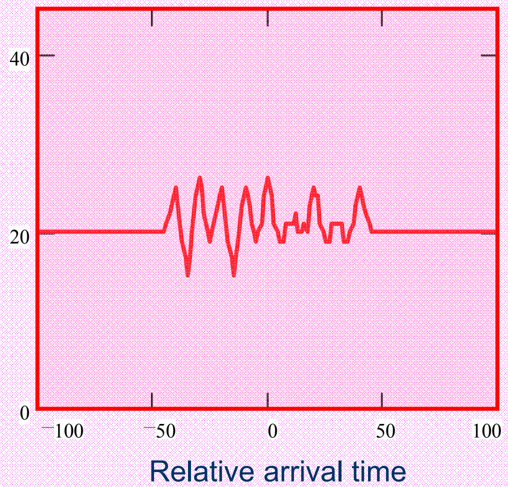




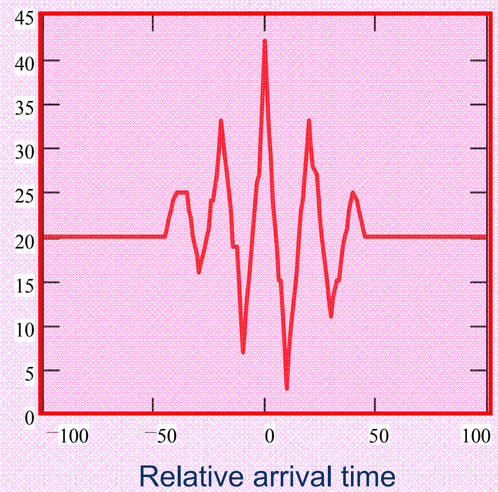
Code coincidence measurement – ***code length: 45 bits***



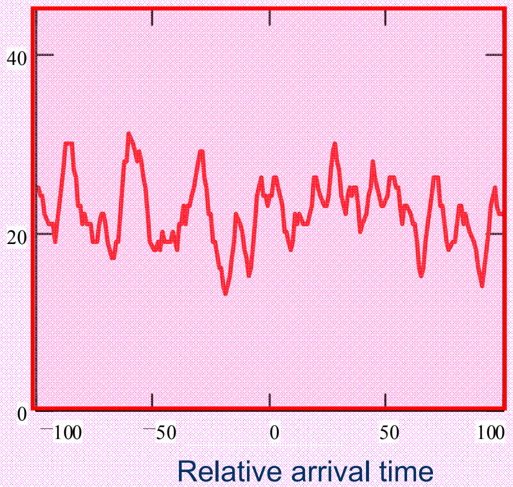
Code comparator output: A with X



Code comparator output: B with X



Code comparator output: C with noise



Codes coincidences: [A xnor X] : 26. ***[B xnor X] : 42,*** [C xnor X] : 28

Question: Is B the transmitted code?

Why can we say that both the remote controller and the receiver are dumb?

The repetition of a same code is taken into account neither by the receiver nor by the remote!

Solution:

To introduce « some statistics » in the receiver

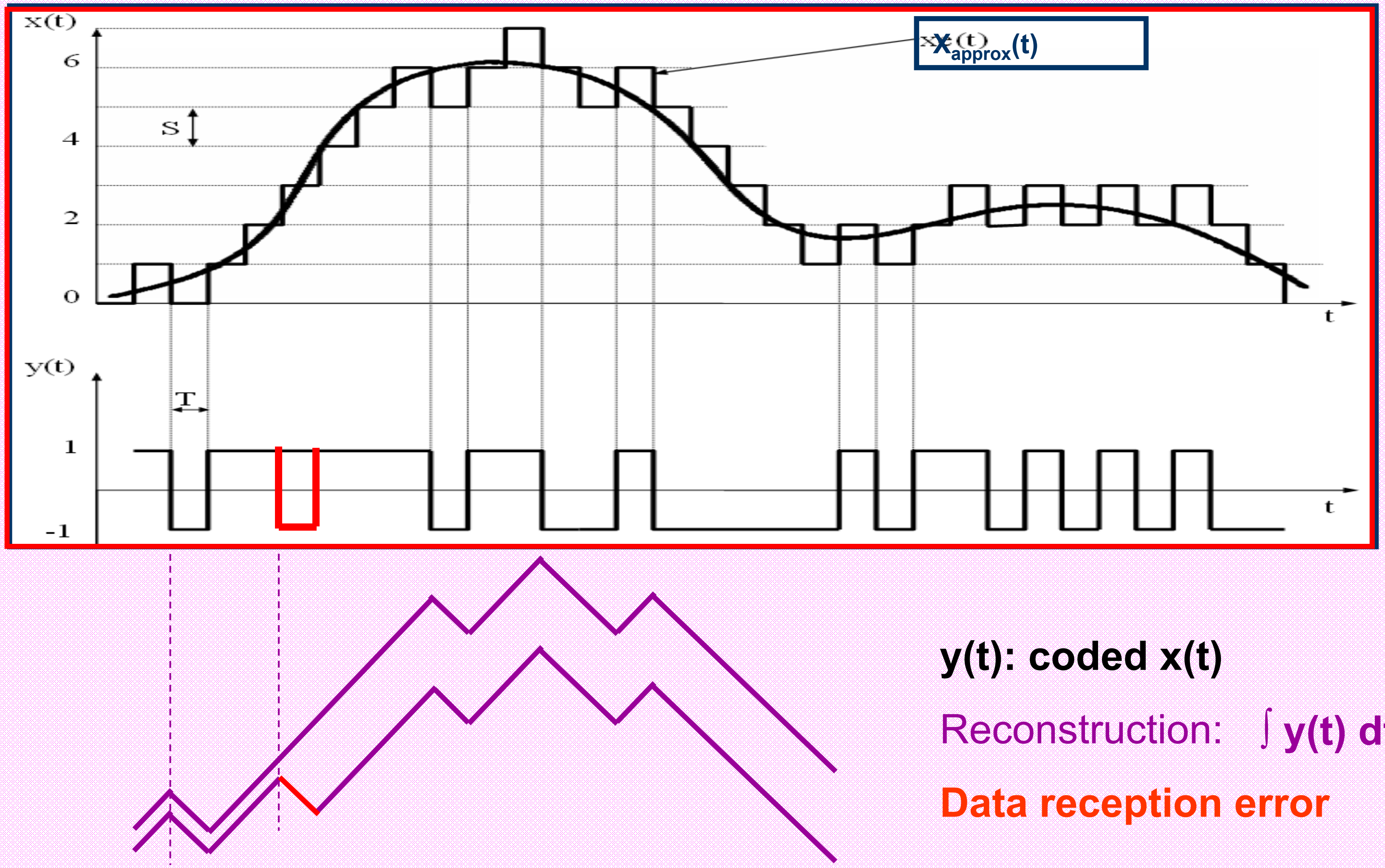
To temporarily increase the remote power

Decision criteria?

What are the consequences of a wrong classification?

E.g. TV remote control
Lighting control
Security

CODING, ENCRYPTION - *Coding example: Delta modulator*

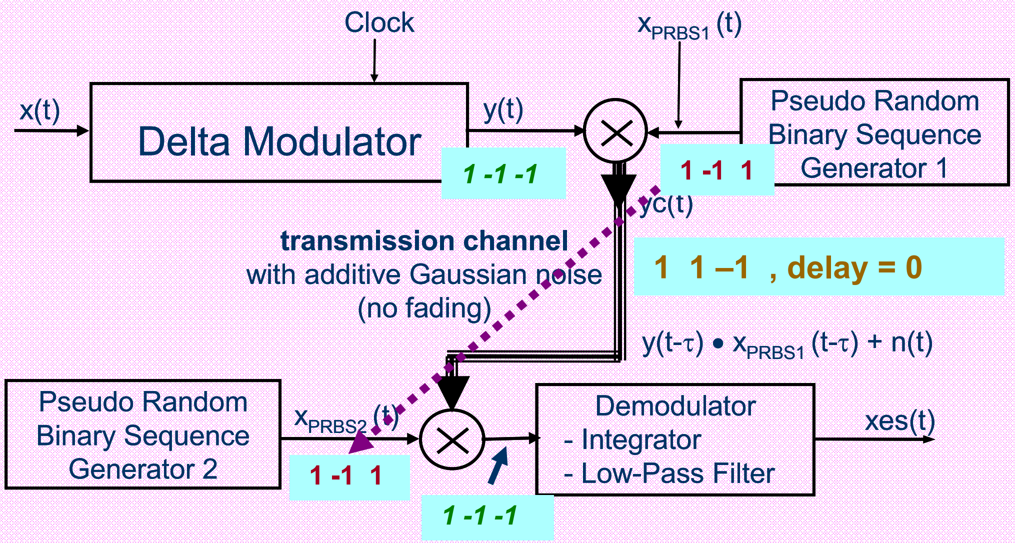


y(t): coded x(t)

Reconstruction: $\int$ **y(t) dt**

Data reception error

CODING, ENCRYPTION - *Scrambling*



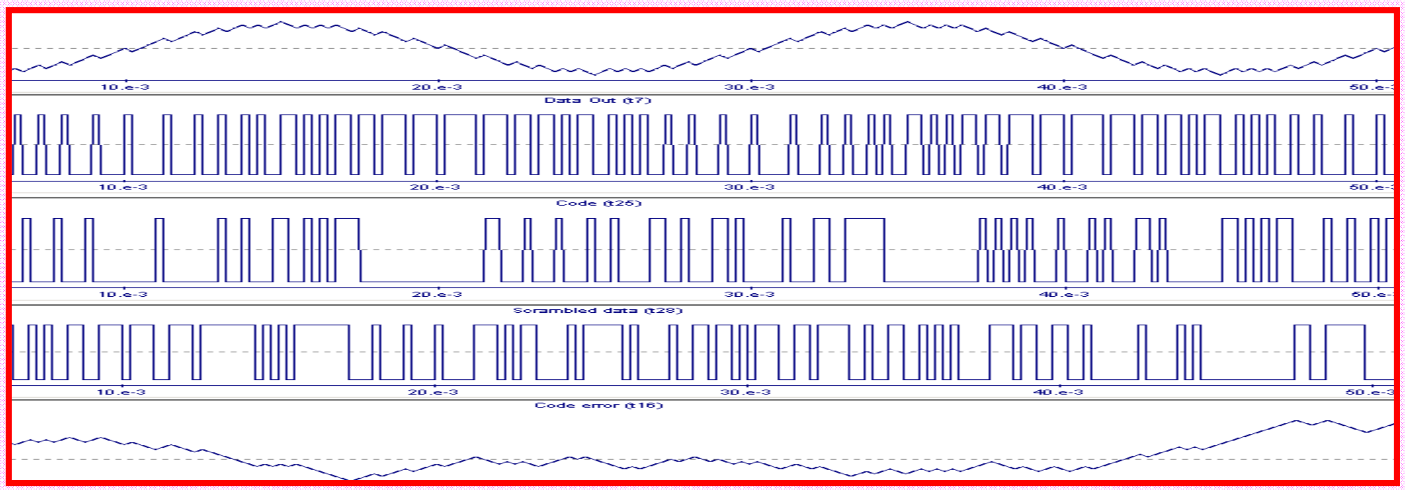
Key issues:

Identical code:

$x_{PRBS1}(t) = x_{PRBS2}(t)$

Synchronization:

Delay – Tracking…



$\int y(t)\,dt$

$y(t)$

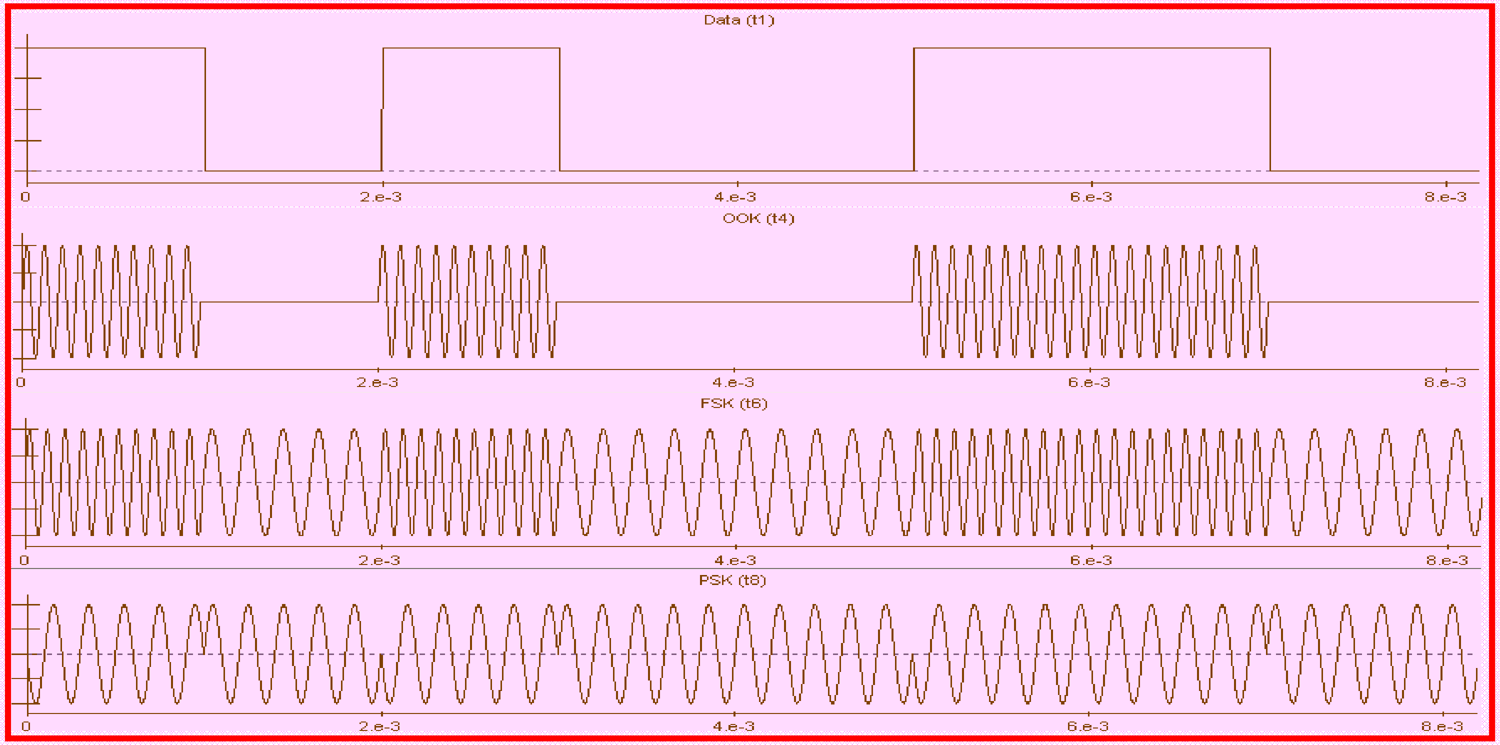
$x_{PRBS1}(t)$

$y(t)\ x_{PRBS}(t)$

$\int y(t)\ x_{PRBS}(t)\,dt$

MODULATION, DEMODULATION

Modulation example: ON-OFF Keying (OOK)
Frequency-shift keying (FSK)
Phase-shift keying (PSK)



Data
1kb/s

OOK
10 kHz

FSK
5 – 10 kHz

PSK
5 kHz 180^0

Data rate = ? OOK freq = ? FSK freq. = ? PSK freq and phase = ?

AMPLITUDE MODULATION and DEMODULATION: Introduction

xi(t) + 1

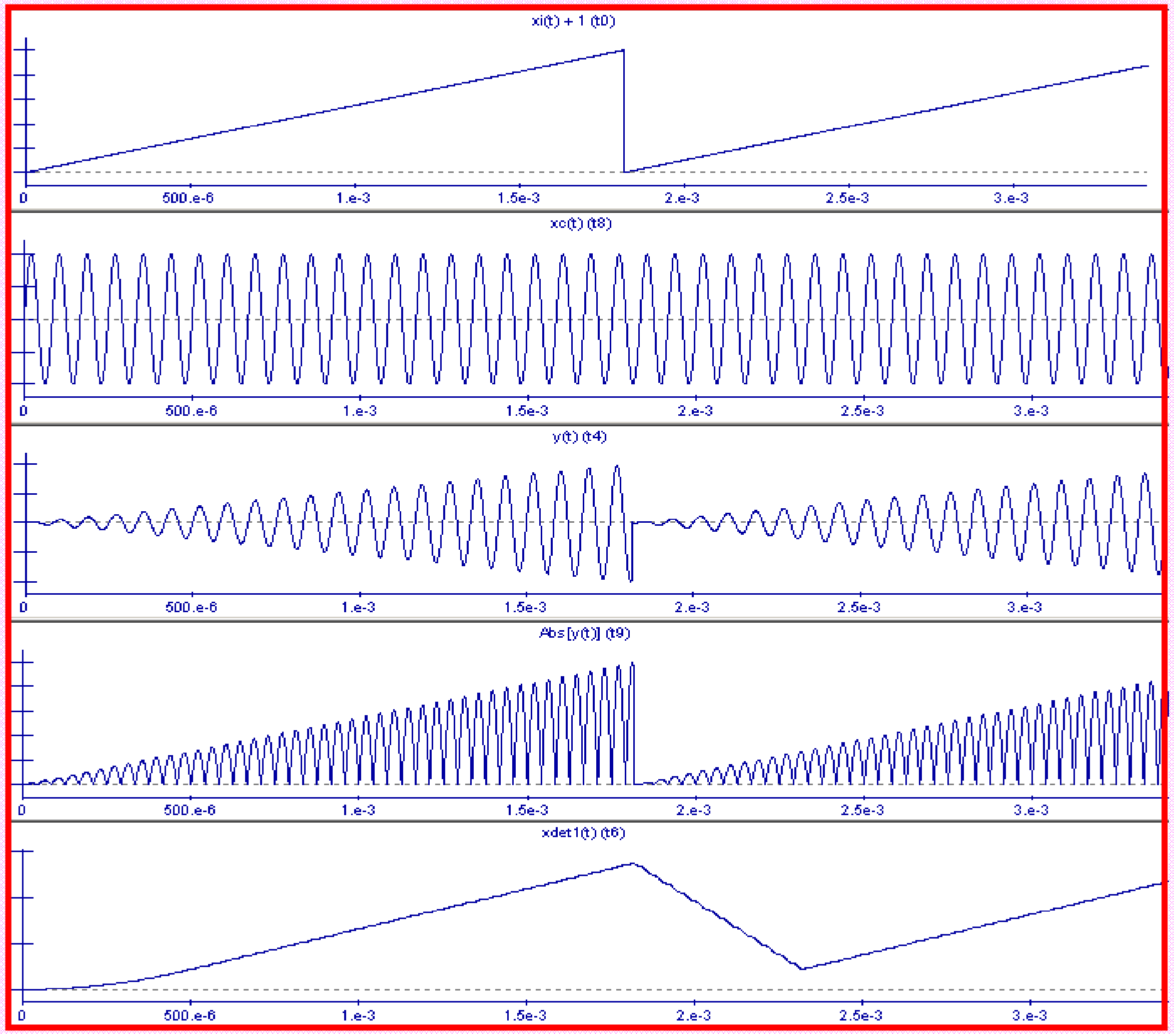
xc(t)

y(t) = [xi(t) + 1] · xc(t)

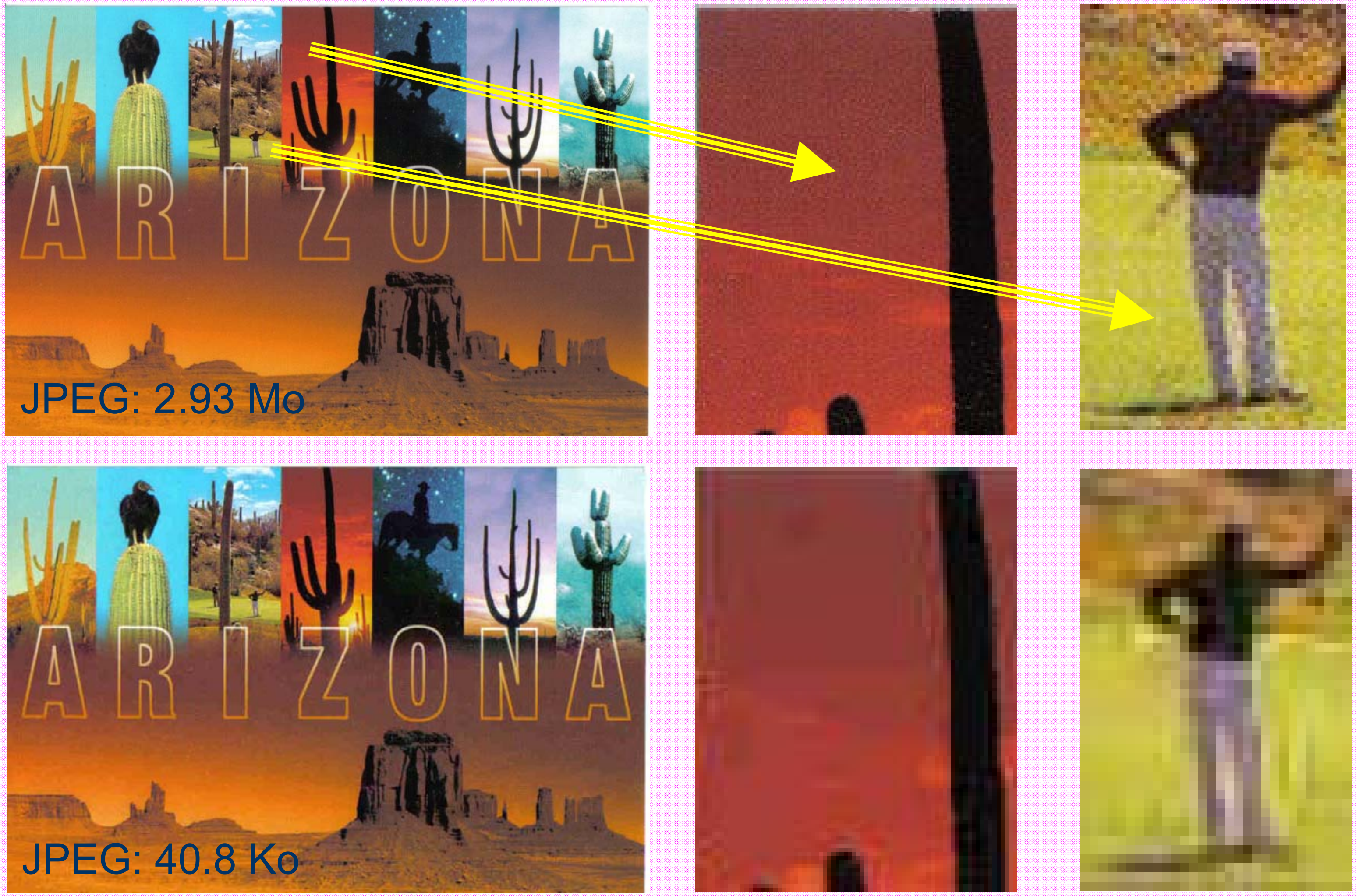
| y(t) |

xdet1(t)

Low-pass filter on | y(t) |

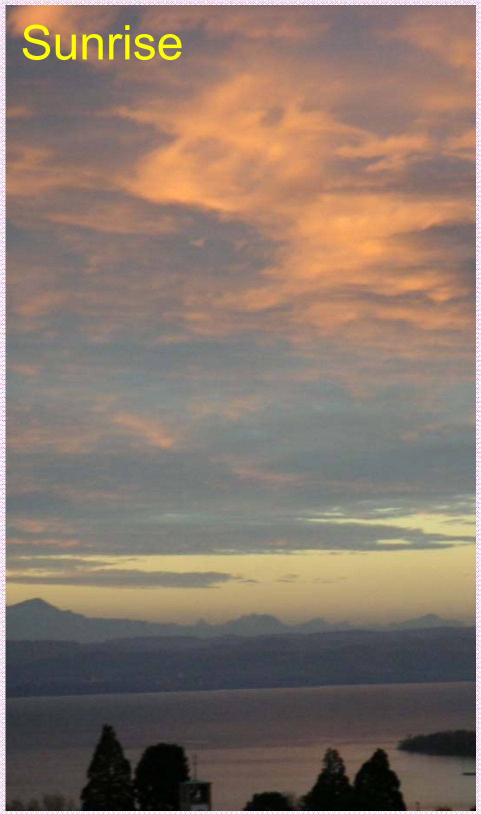


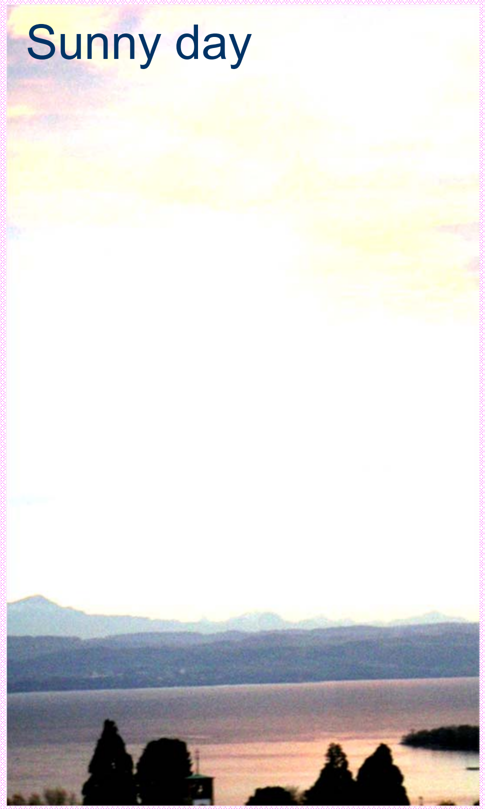
SYNTHESIS , COMPRESSION: Image compression



Analog and Digital Signal Processing

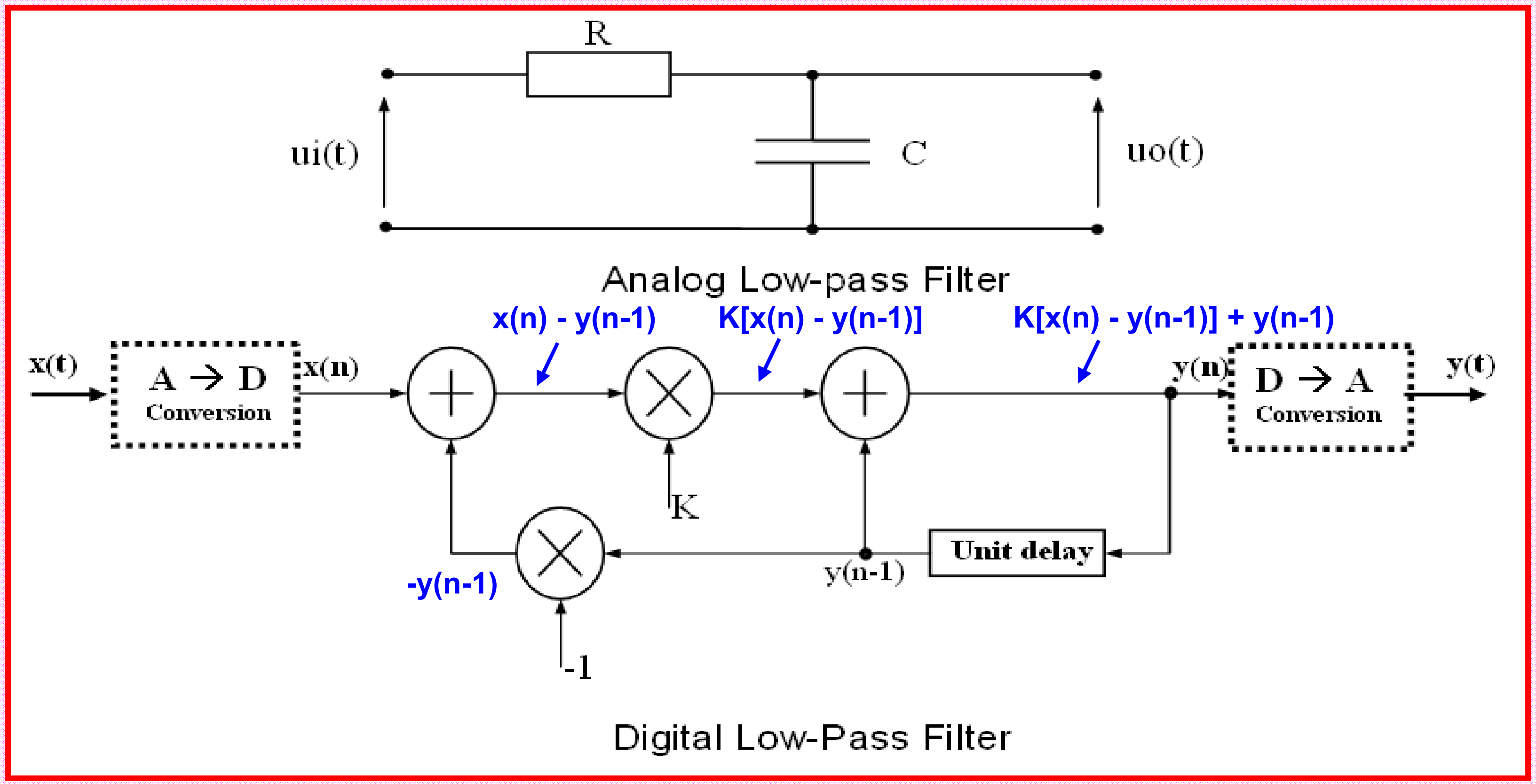
PERCEPTUAL ENHANCEMENT: Image Processing





The choice of the best processing approach is made according to the image content and the desired features we try to emphasize.

ANALOG/DIGITAL SIGNAL PROCESSING



Analog: ui(t) = uo(t) + RC uo'(t)

Digital: y(n) = K x(n) + (1-K) y(n-1)

Unit step response

ui(t) = 1 if $t \geq 0$; ***ui(t) = 0*** if $t < 0$

x(n) = 1 if $n \geq 0$; ***x(n) = 0*** if $n < 0$

Analog:

$uo(t)_{t \geq 0} = 1 - e^{-t/\tau}$, $\tau = 10$

Digital:

y(n) = K x(n) + (1-K) y(n-1)

y(0) = K x(0) + (1-K) y(-1)

y(1) = K x(1) + (1-K) y(0)

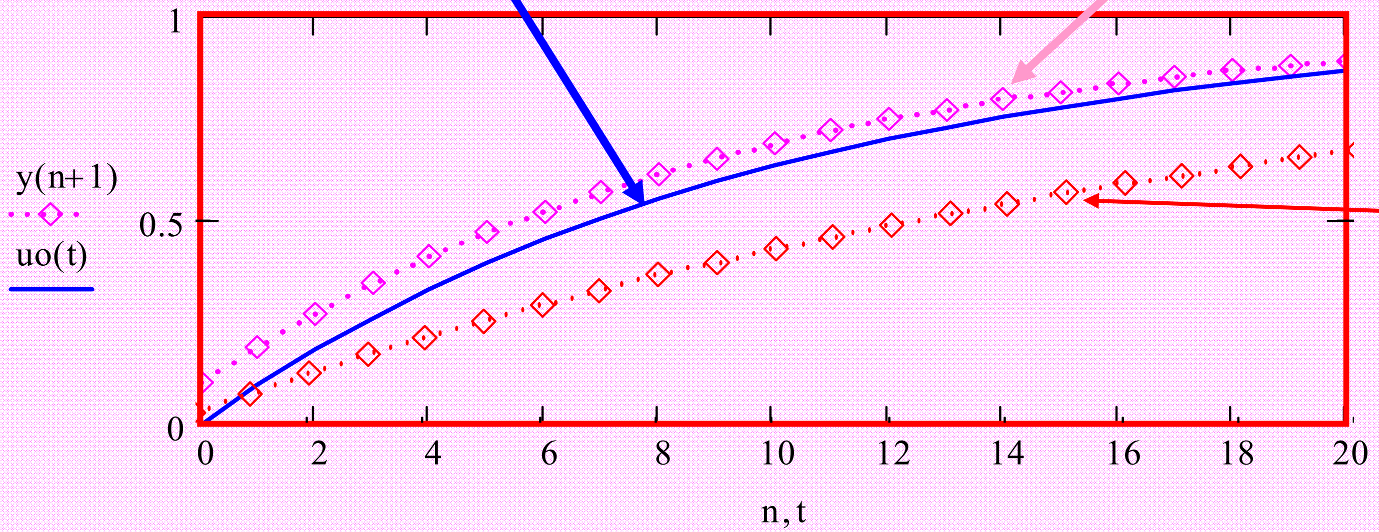
y(2) = K x(2) + (1-K) y(1)

K = 0.1

y(0) = 0.10

y(1) = 0.19

y(2) ≈ 0.27



The step responses are quasi identical!

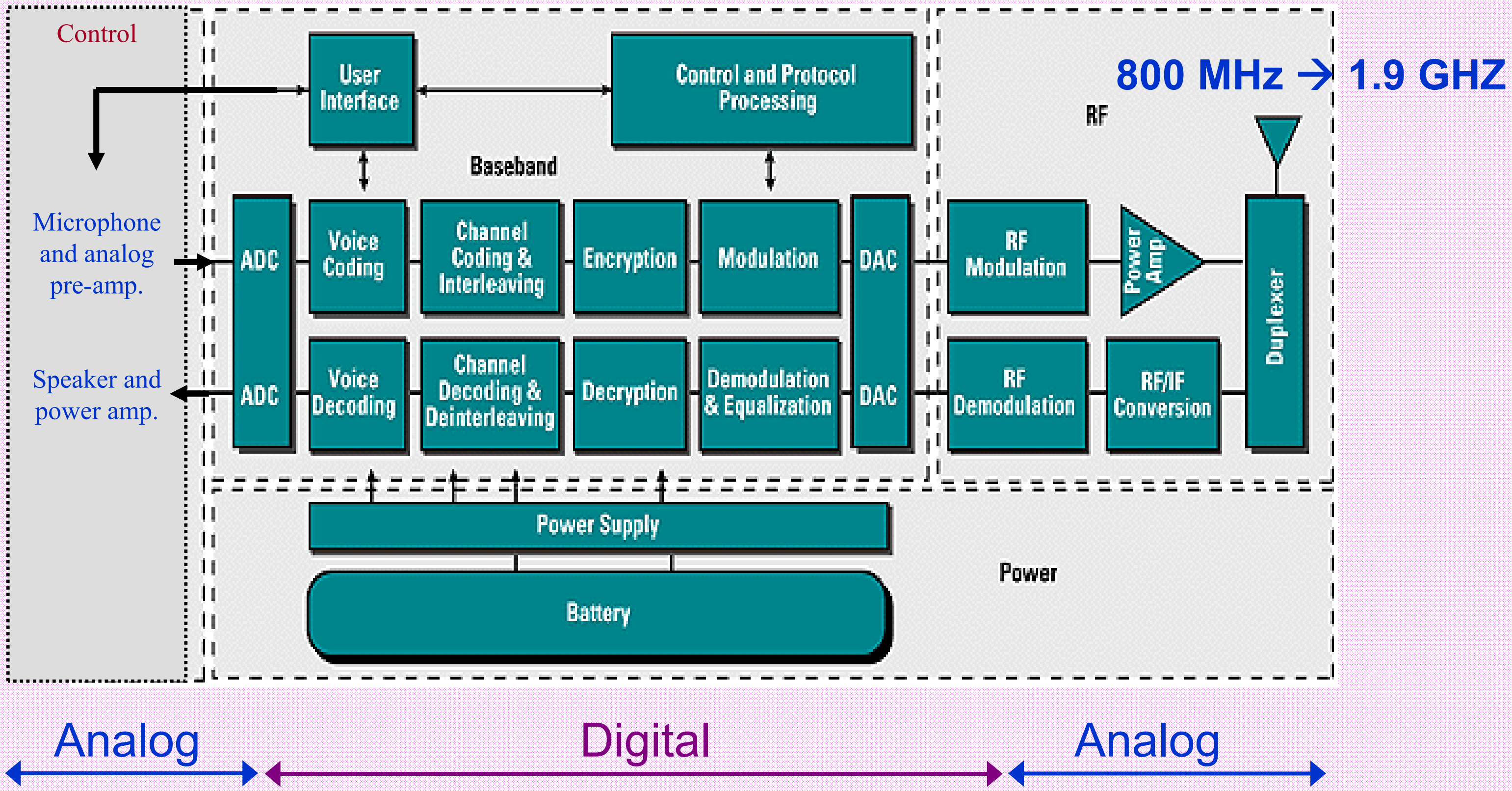
Analog and Digital Signal Processing

COMPARISON between **analog** and **digital** signal processing

- **Repeatability, Long Term Stability**
- **Re-programmability (flexibility)**
- **Adaptation (the algorithms follows the signal characteristics)**
- **Realization of complex non-linear functions**
- **Accuracy (Crystal controlled in DSP cases)**
- **Sensibility to environmental changes (temp., humidity ….)**
- **Speed (maximum operating frequency)**
- **Dynamics range**
- **Power consumption**
- **EMC: self noise and susceptibility**
- **Development time : first time design and redesign**
- **Hardware cost: Complexity and performance dependant**
- **……………………………………………………**

Cell phone:

Portable Wireless Transmitter/Receiver system



All the signal processing goals are embedded in our cell phone!

Problem 1: Estimation, filtering

Consider a length 3 median filter.

a) If ***x(n)***, the input, is the following, determine ***xmedian(n)***, the output of the filter.

n	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17
x(n)	0	0	0	10	10	11	10	18	10	10	0	-1	-10	1	0	1	10	11

b) Compare the median filter performance with a running averager of length 3.

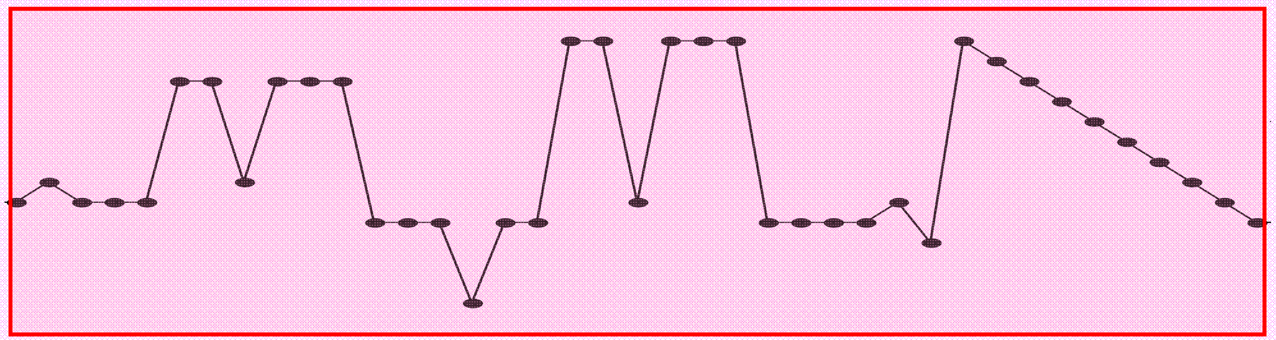
$$y_{run3}(n) = (1/3)\,[x(n) + x(n-1) + x(n-2)]$$

c) What happens if the median filter length is increase from 3 to 5.

d) What is the unit step response of a median filter?

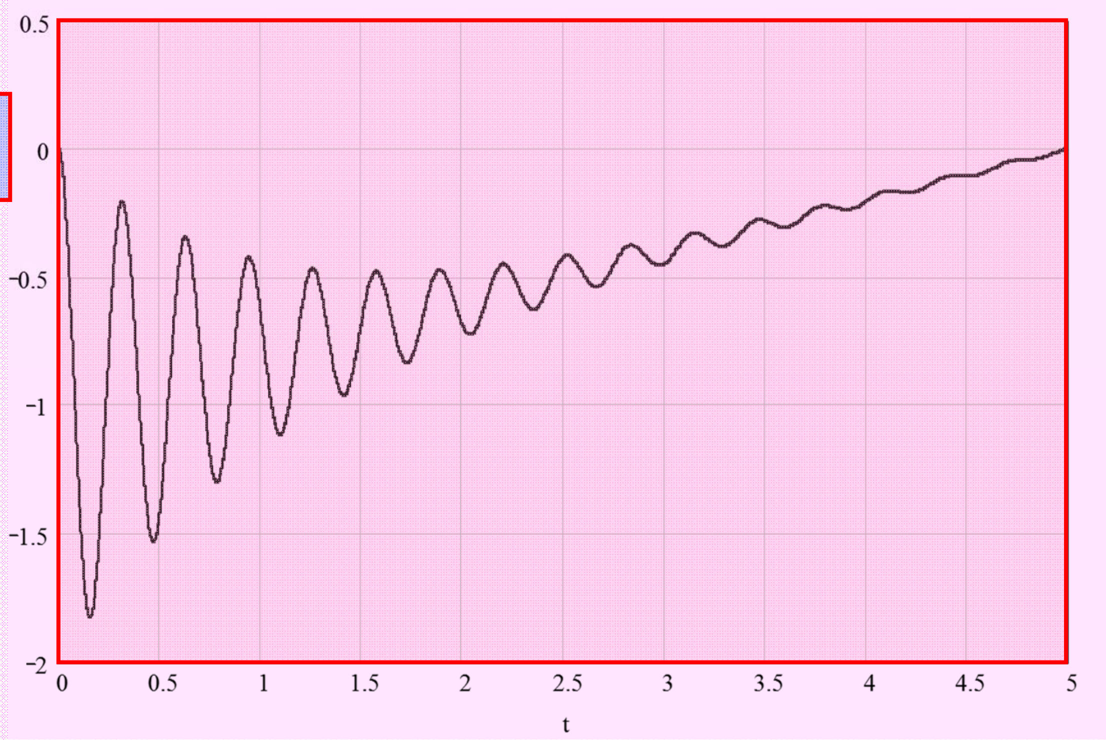
Problème 2: Estimation, filtrage

Tracer l’effet d’un filtre médian et d’un moyenneur glissant (longueur L=3 dans chaque cas) pour le signal suivant :



Problème 3: Modélisation

$$y(t) := \left(A + B \cdot e^{-C \cdot t} \cdot \cos(w_0 \cdot t)\right) + D \cdot t$$



Déterminer ***approximativement***

A, B, C, D et ω_0

Procédure :

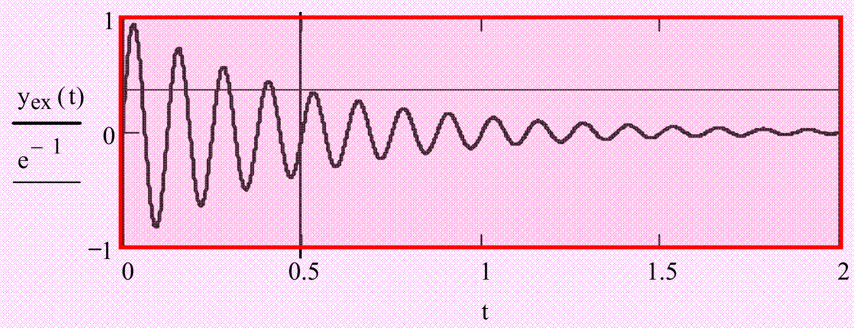
1) Déterminer ω_0

2) Estimer où passe la droite ***D·t*** afin d'évaluer ***D***.

3) Considérer ***t=0*** → etc ……..

Rappel :

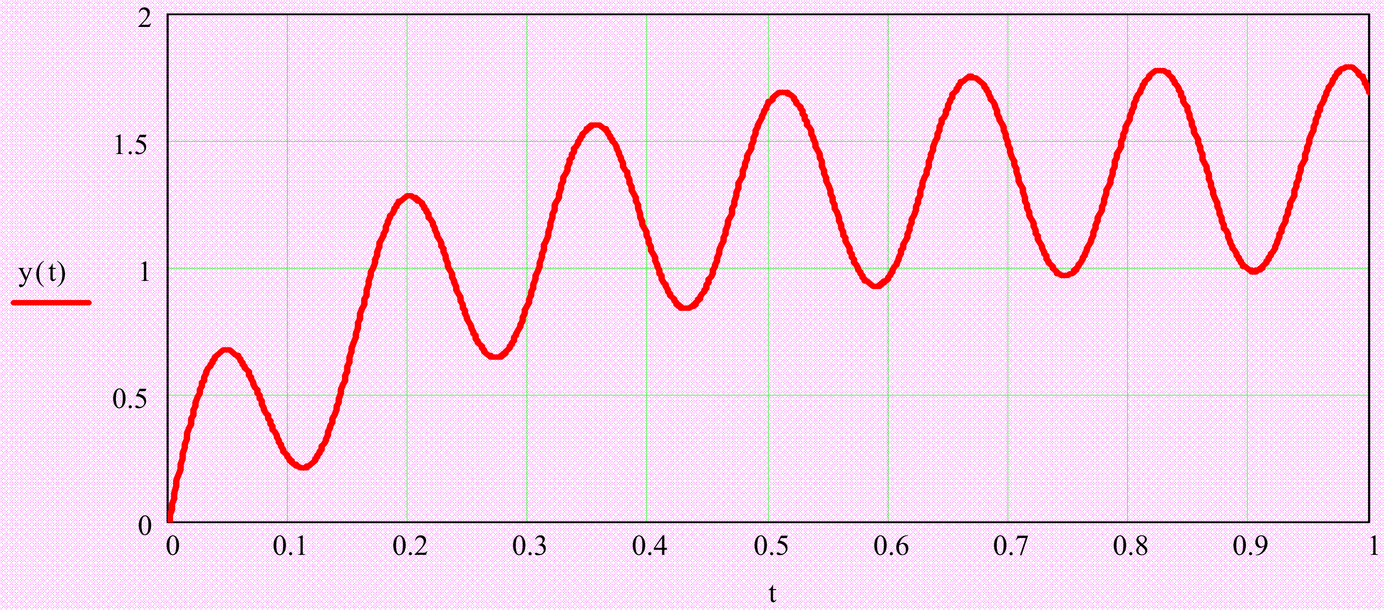
$$y_{ex}(t) := e^{-2 \cdot t} \cdot \sin(50 \cdot t)$$



$e^{-1} \approx 0.37$ $\omega_0 = 2\,\pi\, f_0 = 2\,\pi\,/\,T$

Problème 4: Modélisation

Réponse d'un "système" à une excitation :



$$y(t) = \alpha \cdot \left(1 - e^{\frac{-t}{\tau}}\right) + \beta \cdot \sin(\omega_o \cdot t)$$

Déterminer chacun de ses paramètres.

α = ? **β = ?**

τ = ? **ω_o = ?**

TiePie engineering

Handyscope HS5

TiePieSCOPE HS805

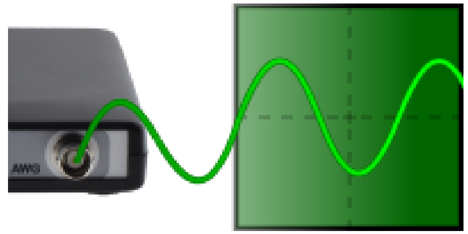
Handyprobe HP3

Handyscope HS4

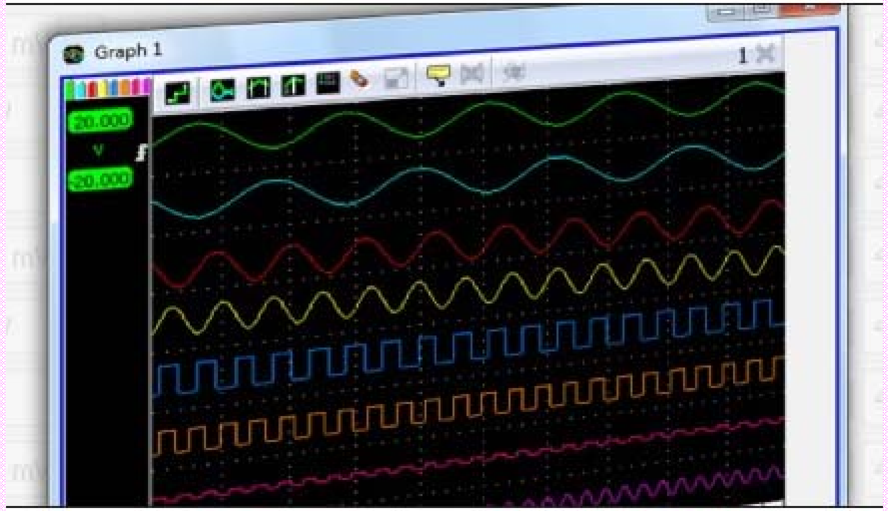
Handyscope HS4 DIFF

Handyscope HS3

Function generators



High quality arbitrary waveform generators for accurate signal generation.



Installation

HS3

Acquisition system	
Number of input channels	2 analog, BNC
Resolution	8 bit ≤ 100 MS/s 12 bit ≤ 50 MS/s 14 bit ≤ 3.125 MS/s 16 bit ≤ 195 kS/s
Accuracy	0.2% ± 1 LSB
Ranges	200 mVolt .. 80 Volt full scale
Coupling	AC/DC
Impedance	1 MOhm / 30 pF
Protection (in all ranges)	200 Volt (DC + AC peak < 10 kHz)
Bandwidth (-3dB)	DC to 50 MHz maximum
AC coupling cut off frequency (-3dB)	1.5 Hz
Maximum sampling rate	100 MS/s, 10 nsec (model HS3-100) 50 MS/s, 20 nsec (model HS3-50) 25 MS/s, 40 nsec (model HS3-25) 10 MS/s, 100 nsec (model HS3-10) 5 MS/s, 200 nsec (model HS3-5)
Sampling source	internal, external
Sampling source internal	quartz
Accuracy	±0.01%
Stability	±100 ppm over -40°C to +85°C
Memory	131072 samples each channel

Order code	Max. sampling speed	Price
HS3-AWG-100	100 MS/s	€ 1,148.00
HS3-AWG-50	50 MS/s	€ 1,098.00
HS3-AWG-25	25 MS/s	€ 942.00
HS3-AWG-10	10 MS/s	€ 820.00
HS3-AWG-5	5 MS/s	€ 679.00

www.Tiepie.com

Installation

Arbitrary Waveform Generator (independent from acquisition system)	
Number of output channels	1 analog, BNC
Resolution	14 bit @ 50 MS/s
Amplitude	-12 Volt .. 12 Volt
Accuracy	0.4%
Amplitude step	0 - ±0.1 V range, 8192 steps ±0.1 - ±0.9 V range, 8192 steps ±0.9 - ±12 V range, 8192 steps
Coupling	DC
Impedance	50 Ohm
Bandwidth	DC to 2MHz
DC level	0 - ±12 V in 8192 steps
Noise level	0 - ±0.1 V range : 900 µVolt RMS ±0.1 - ±0.9 V range : 1.3 mVolt RMS ±0.9 - ±12 V range : 1.5 mVolt RMS
Sampling rate	50 MHz
Sampling source	internal
Accuracy	±0.01%
Stability	±100 ppm over -40°C to +85°C
Memory	1024 points DDS mode 128K points linear mode
Waveforms	sine, triangle, square, DC, noise and user defined
Symmetry	1 - 99%, 1% steps

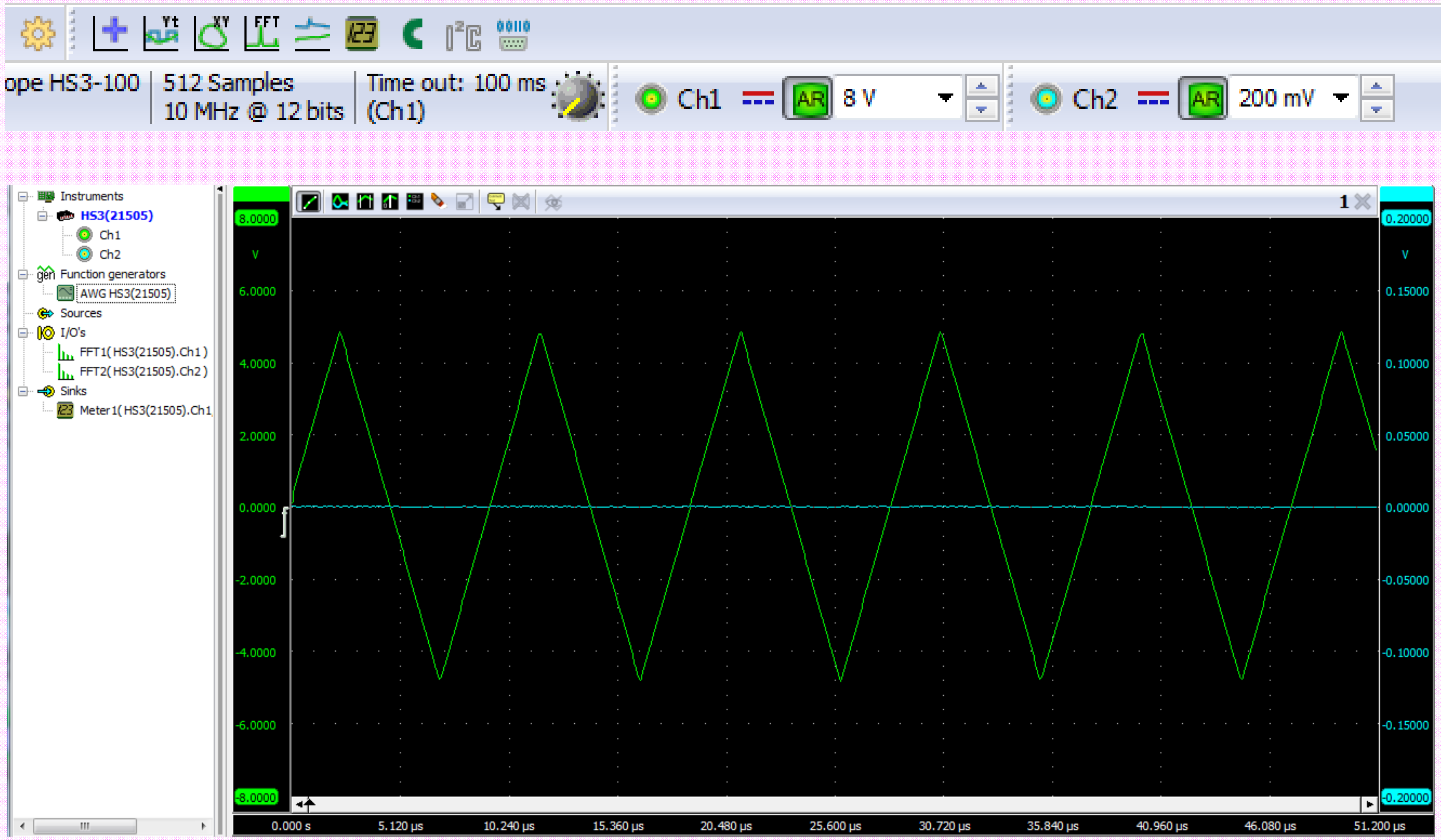
HS3

Trigger system	
System	digital, 2 levels
Source	CH1, CH2, AND, OR, digital external
Trigger modes	rising slope, falling slope, inside window, outside window, peak
Level adjustment	0 - 100% of full scale
Hysteresis adjustment	0 - 100% of full scale
Resolution	0.025% (12 bits)
Digital trigger input level	0 - 3.3 volt (5 volt max)
Pre trigger	0 - 131071 samples (0 - 100%, one sample resolution)
Post trigger	0 - 131071 samples (0 - 100%, one sample resolution)

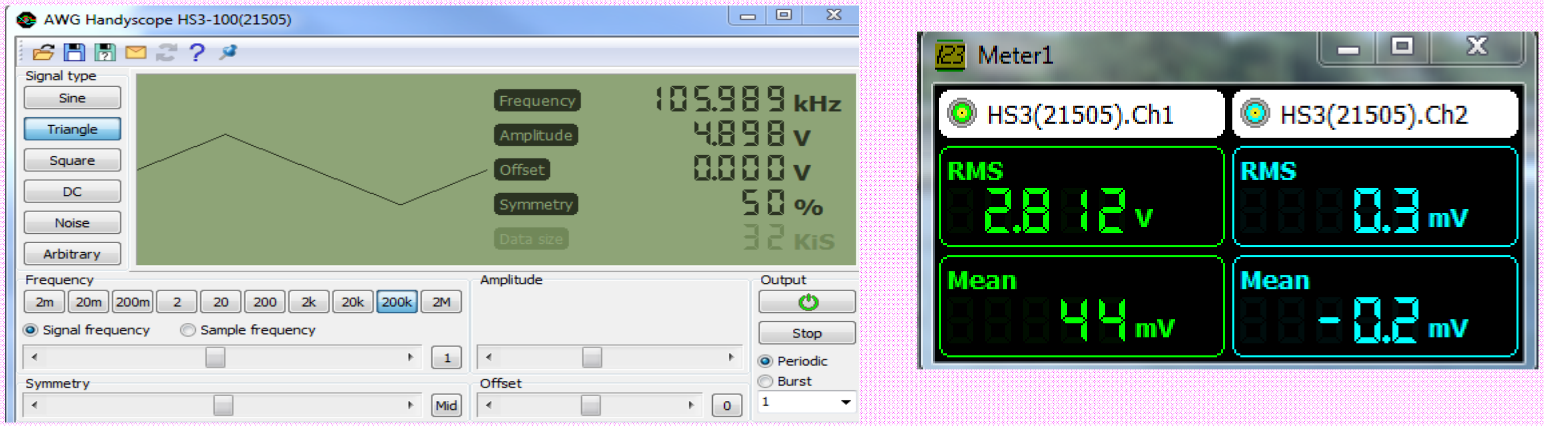
Interface	
Interface	USB 2.0 High Speed (480 Mbit/s); (USB 1.1 Full Speed (12 Mbit/sec) compatible)

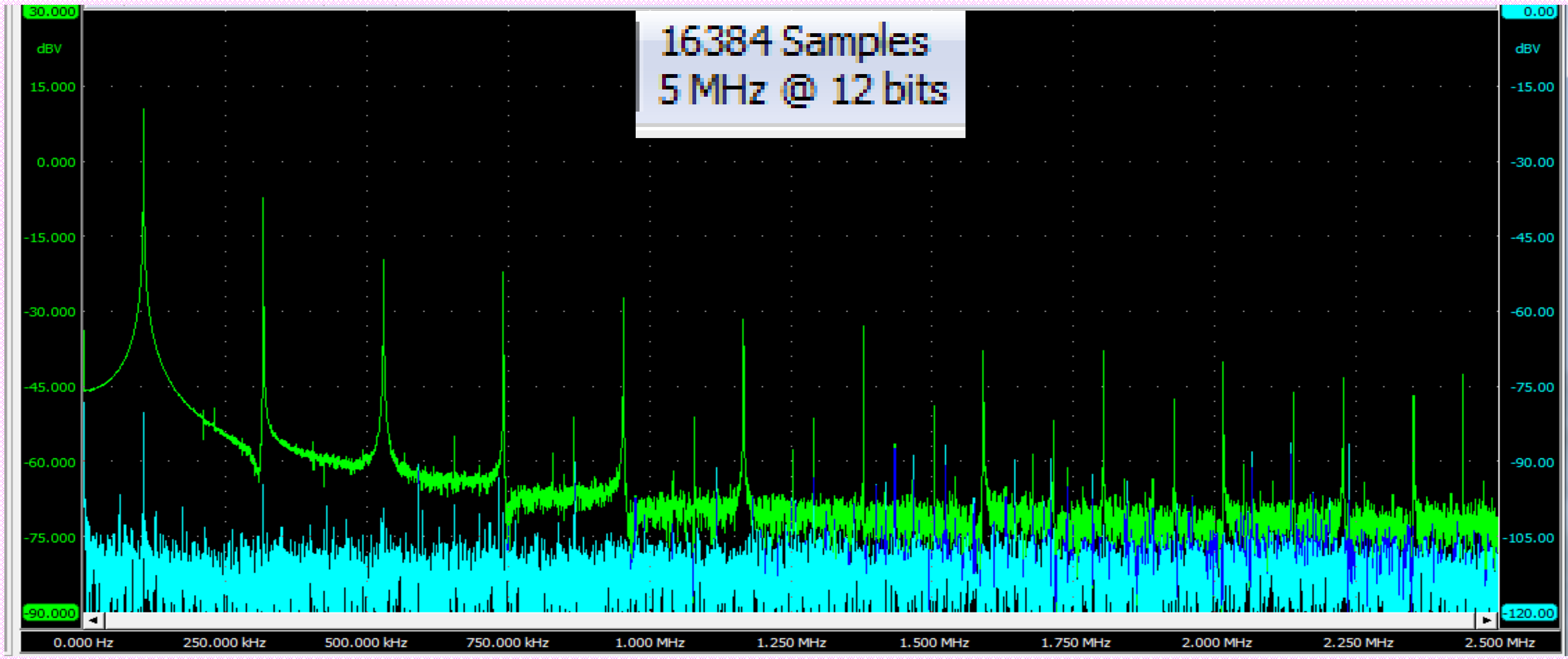
Power Requirements	
Power from USB port	500 mA max (2.5 Watt max)
Power via external power input / extension connector	1500 mA max (7.5 Watt max)
Minimum voltage	4.5 Volt DC
Maximum voltage (SN# <12941)	6 Volt DC
Maximum voltage (SN# >12941)	12 Volt DC

Multi Channel Software (a)

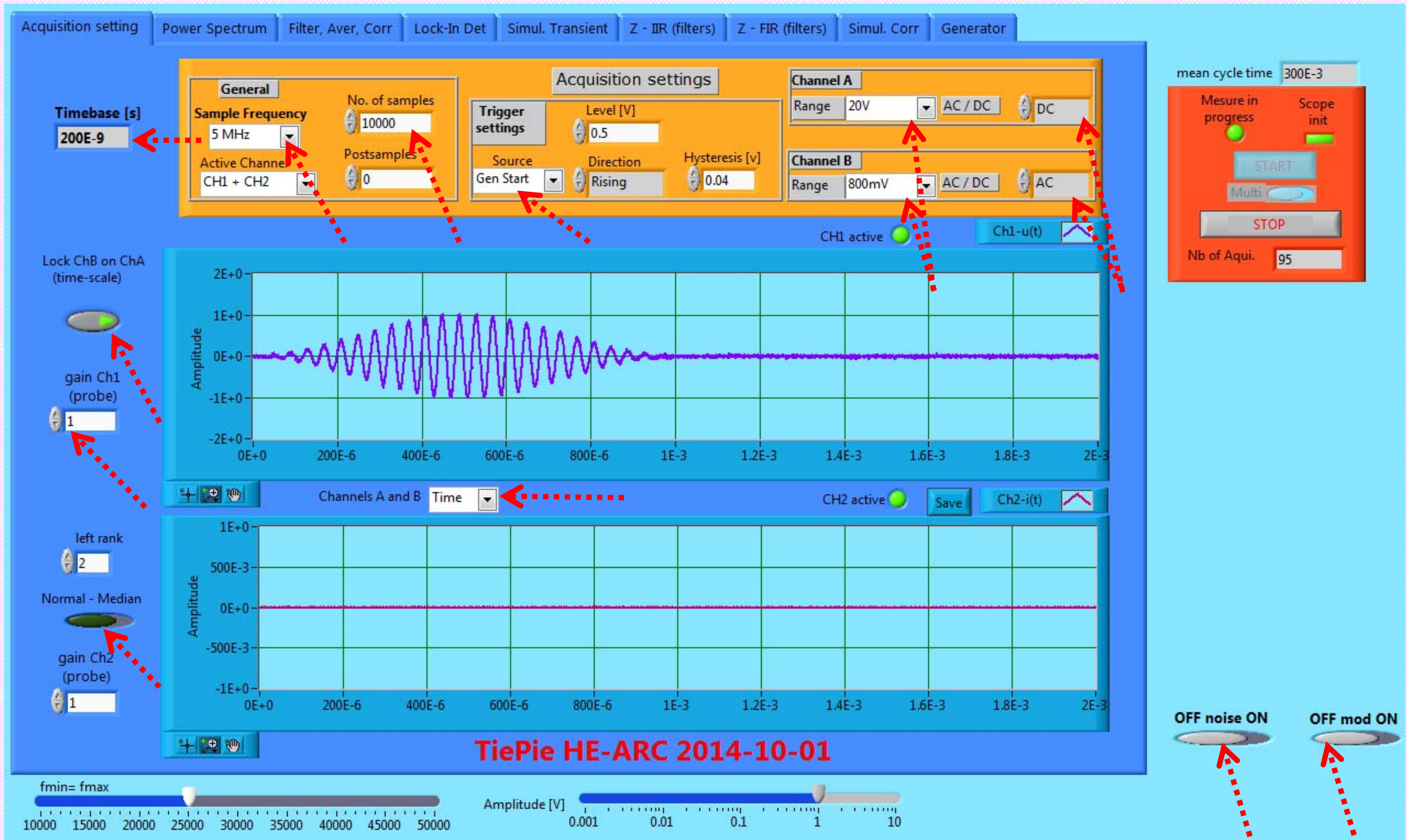


Multi Channel Software (b)

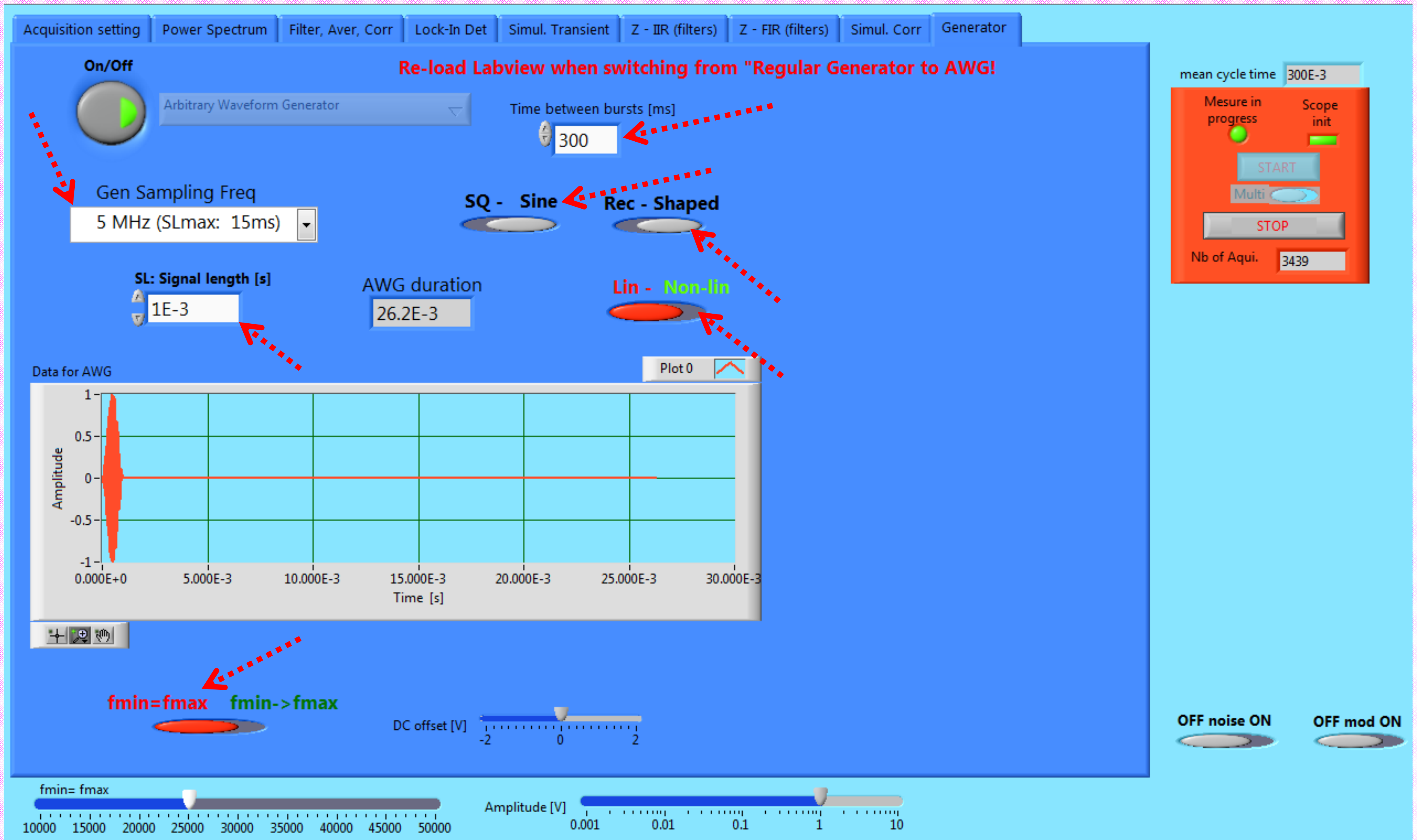




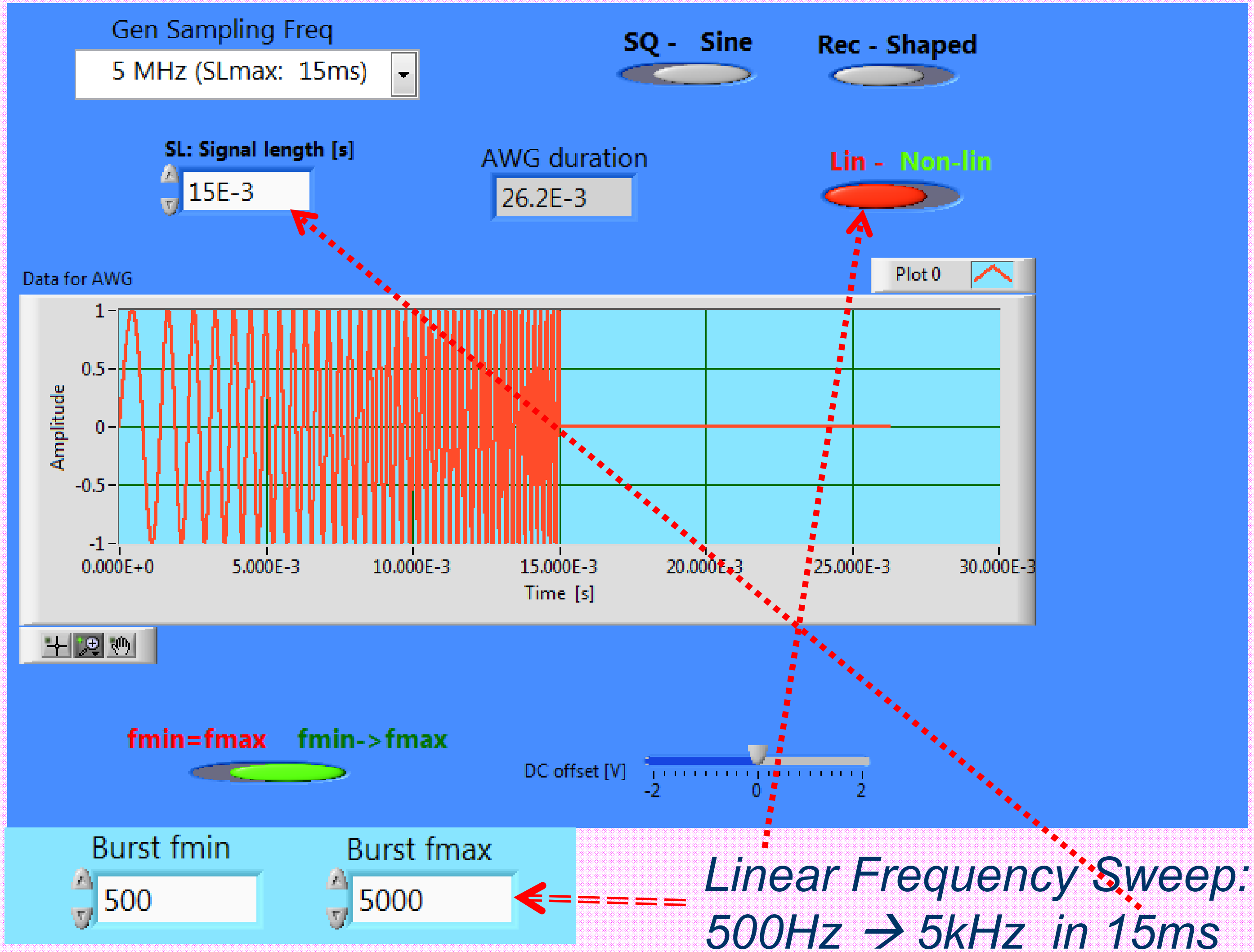
Acquisition settings



Generator (basic 1)

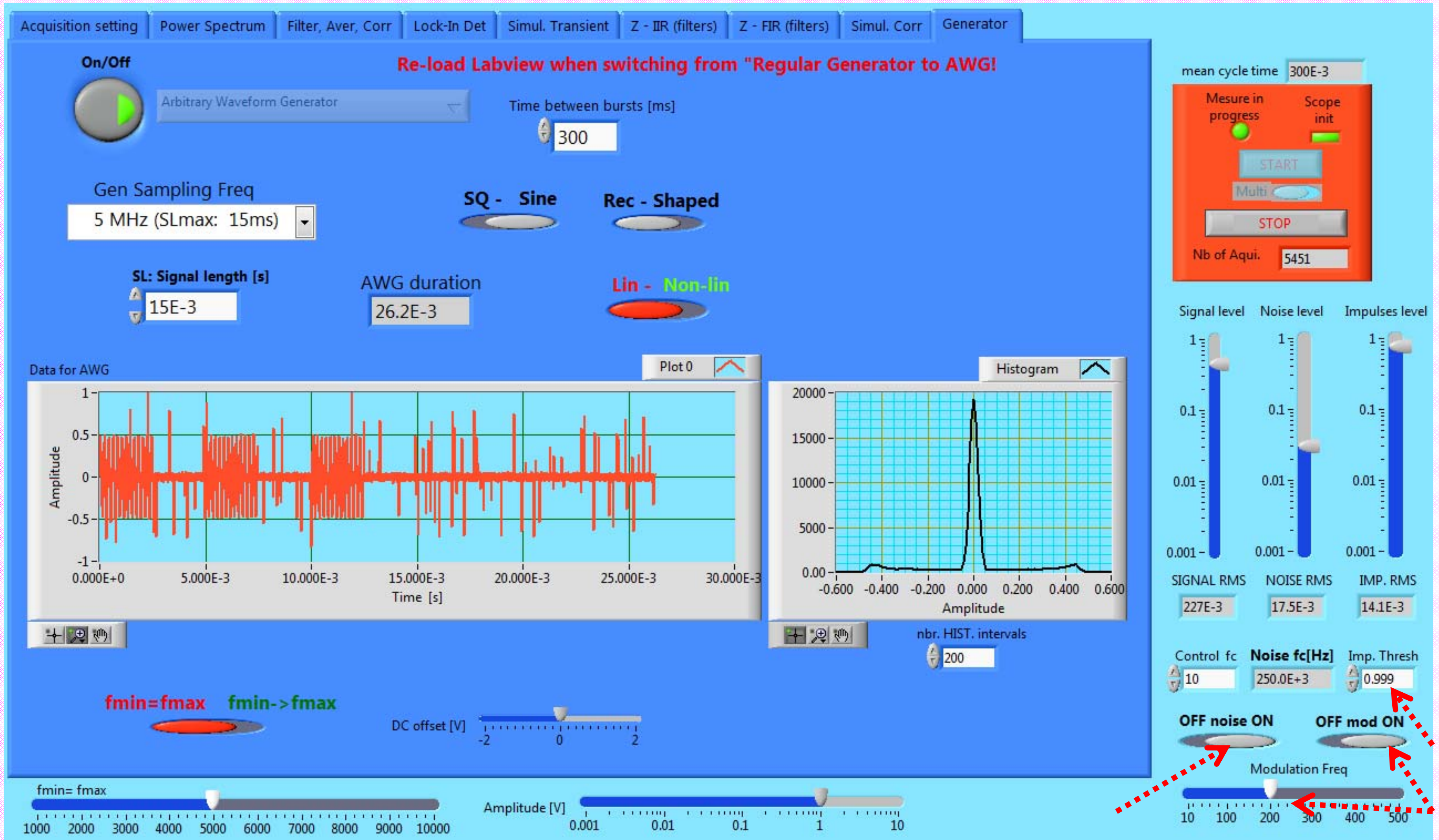


Generator (basic 2)

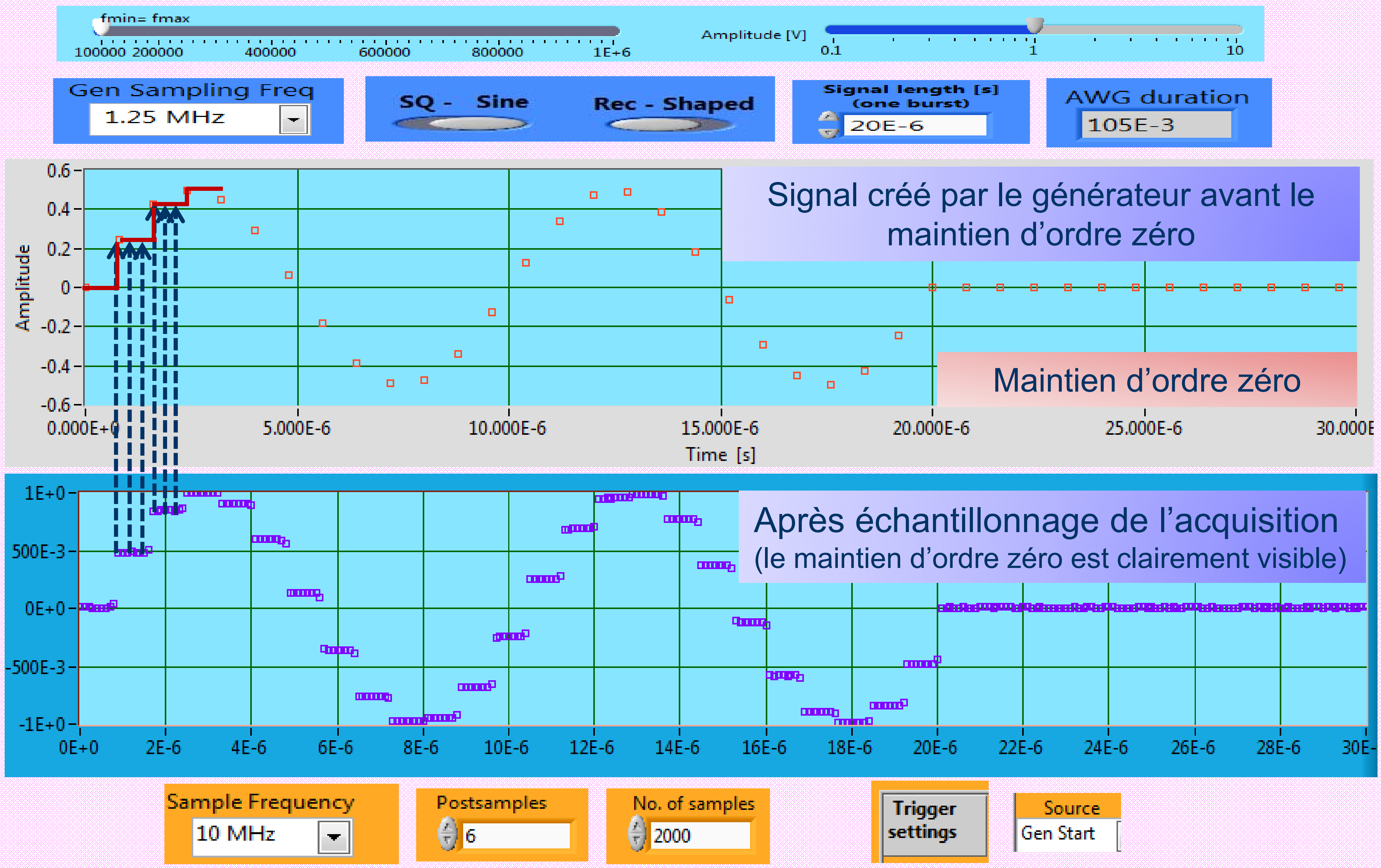


Linear Frequency Sweep:
500Hz → 5kHz in 15ms

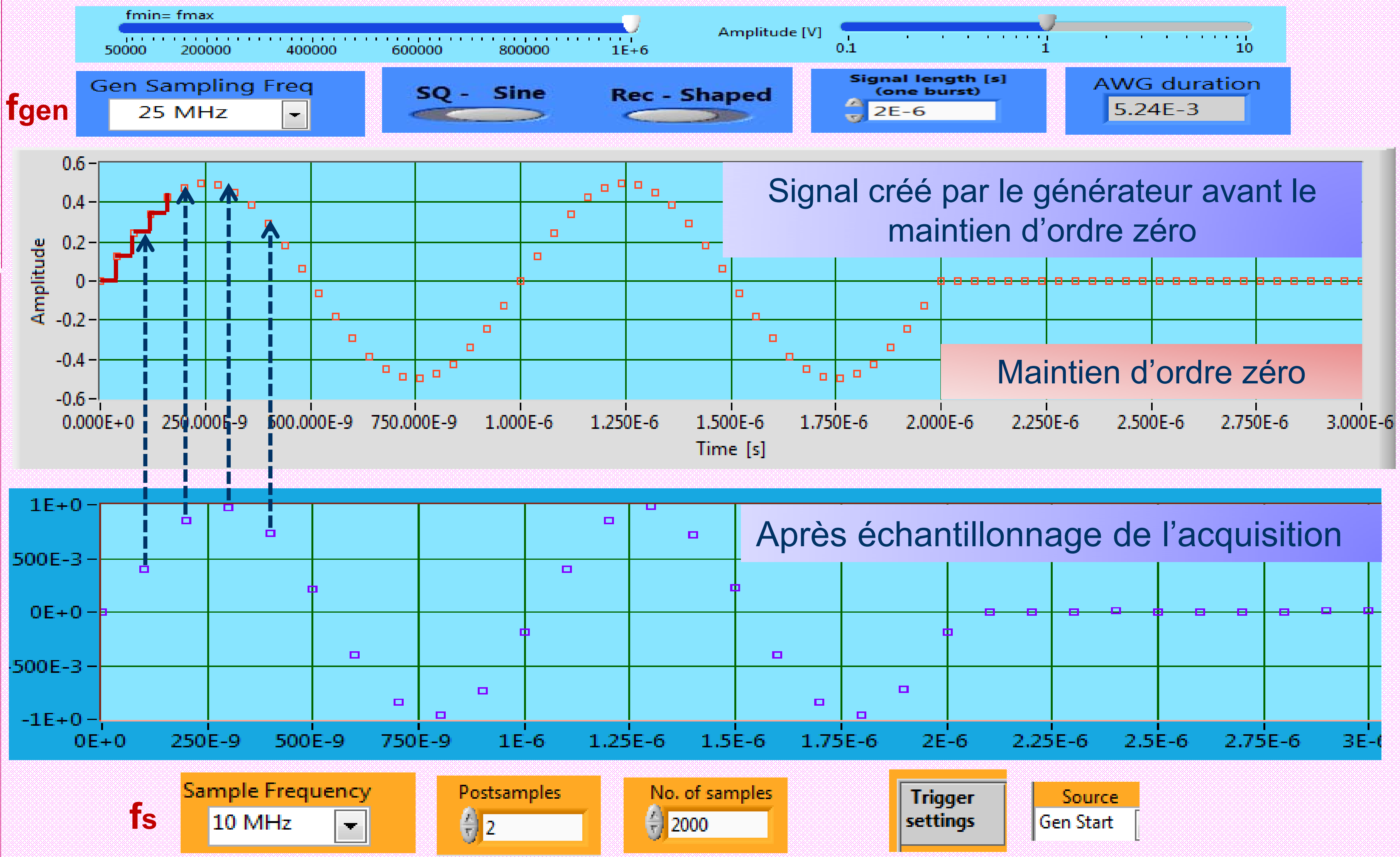
Generator (advanced)



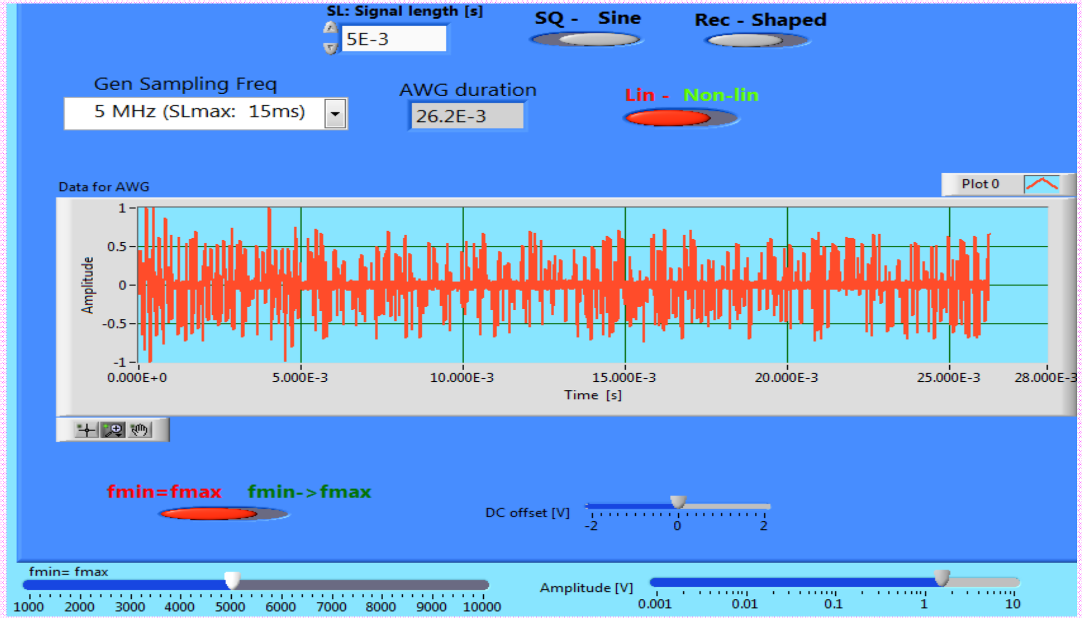
Generator + Acquisition: fgen < fs

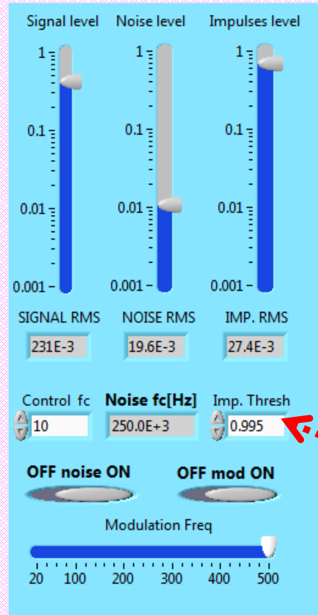


Generator + Acquisition: fgen > fs



Filtre médian



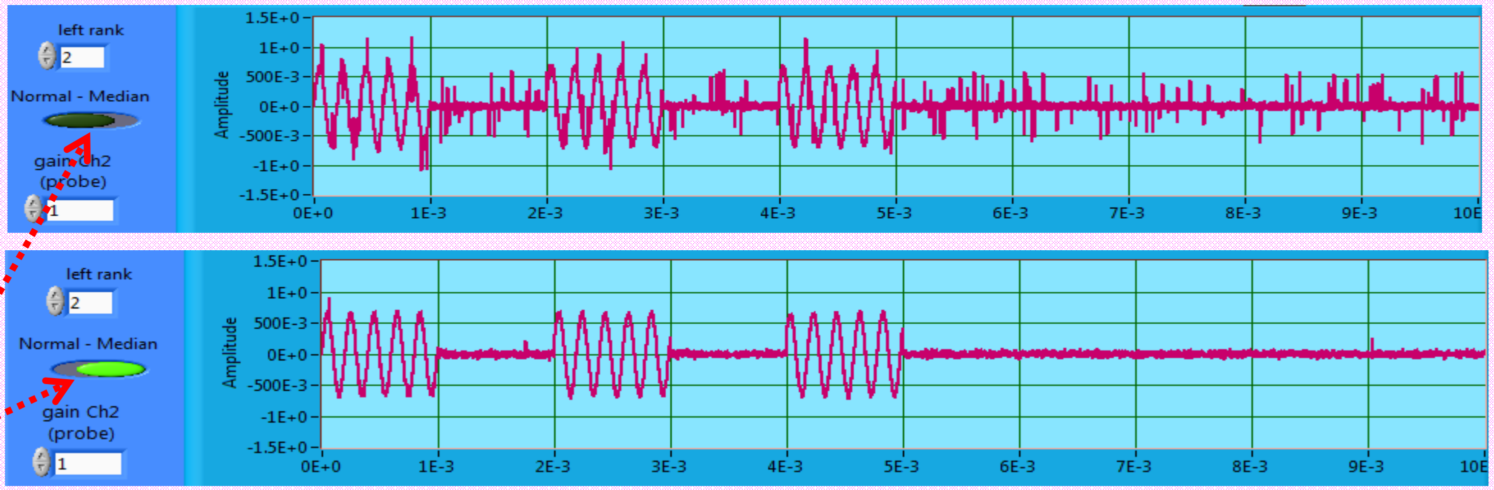


0.995

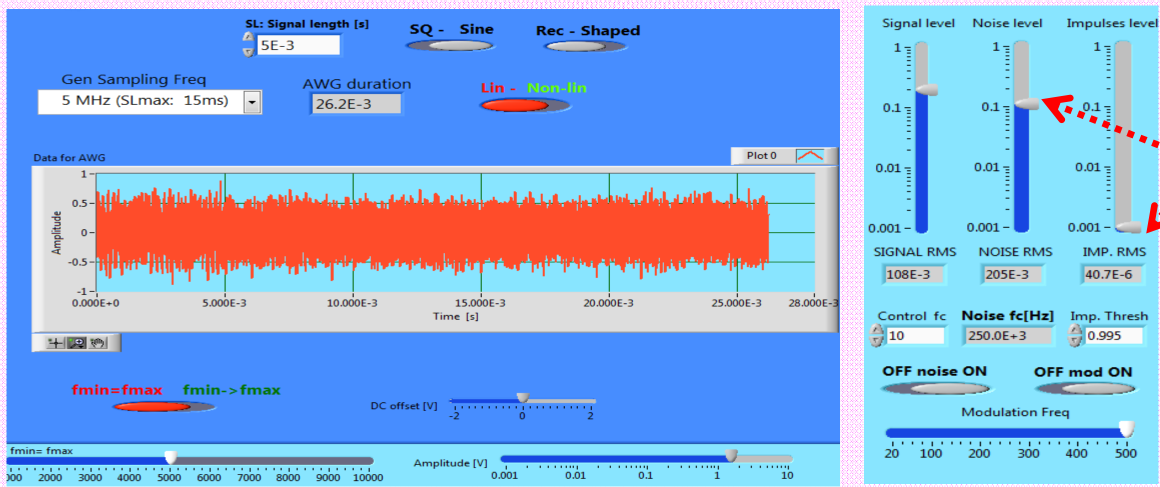
Acquisition

Sample Freq.
5MHz

No. of samples:
50000



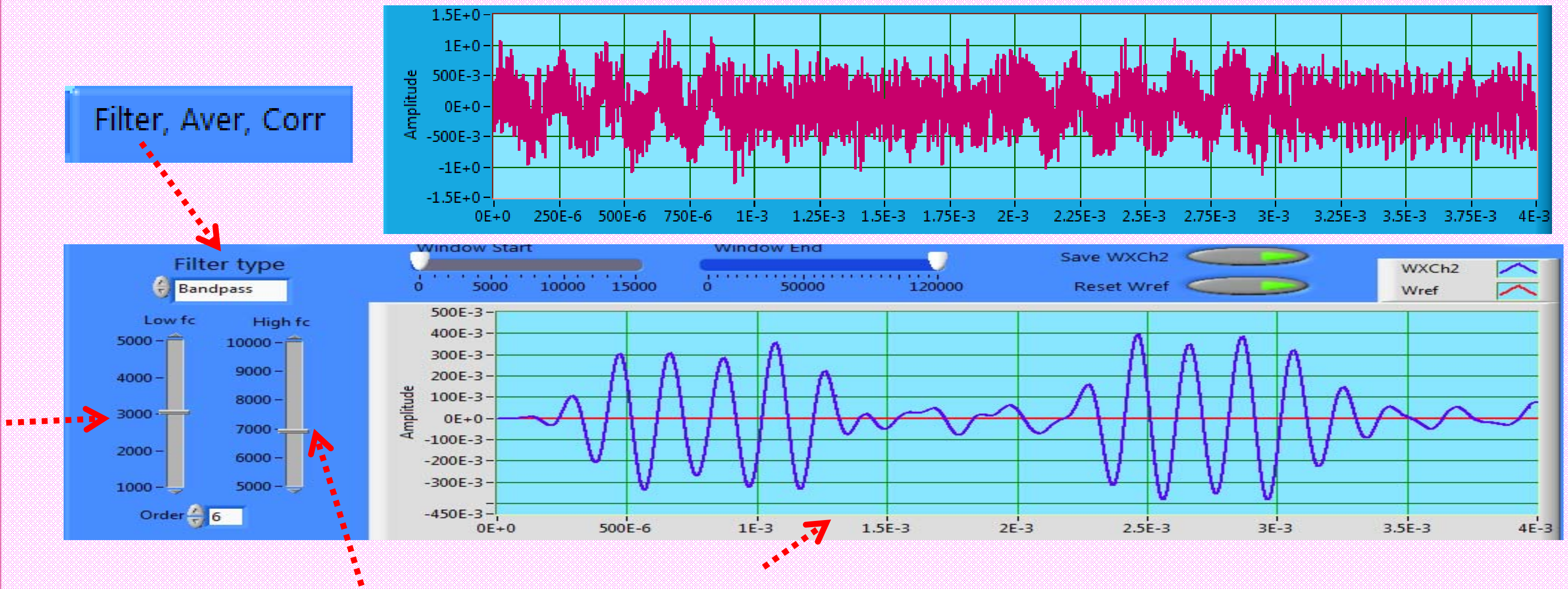
Filter, Aver, Corr



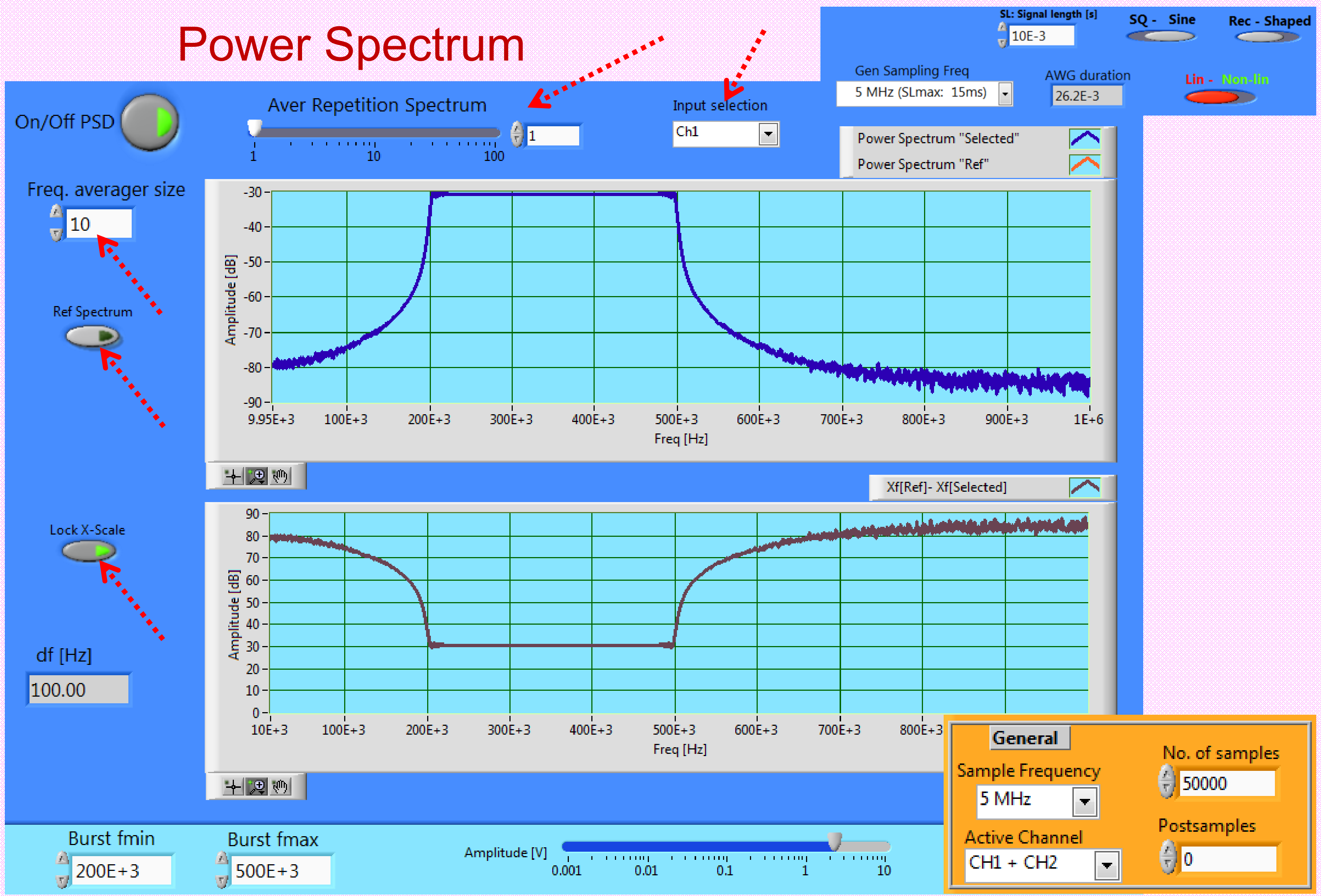
Acquisition Settings

Sample Freq.
5MHz

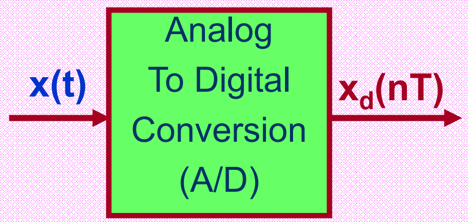
No. of samples:
20000

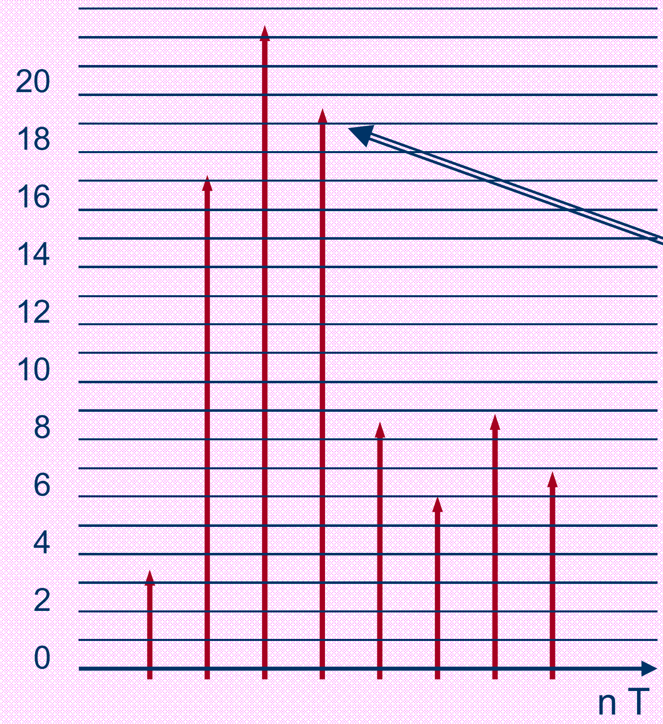


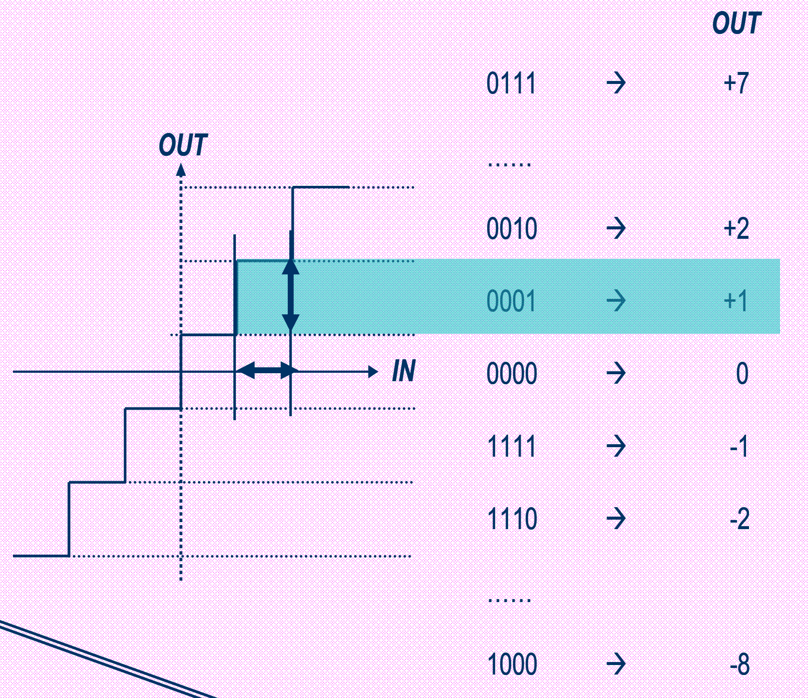
Power Spectrum



Acquisition de données : ***Quantification***







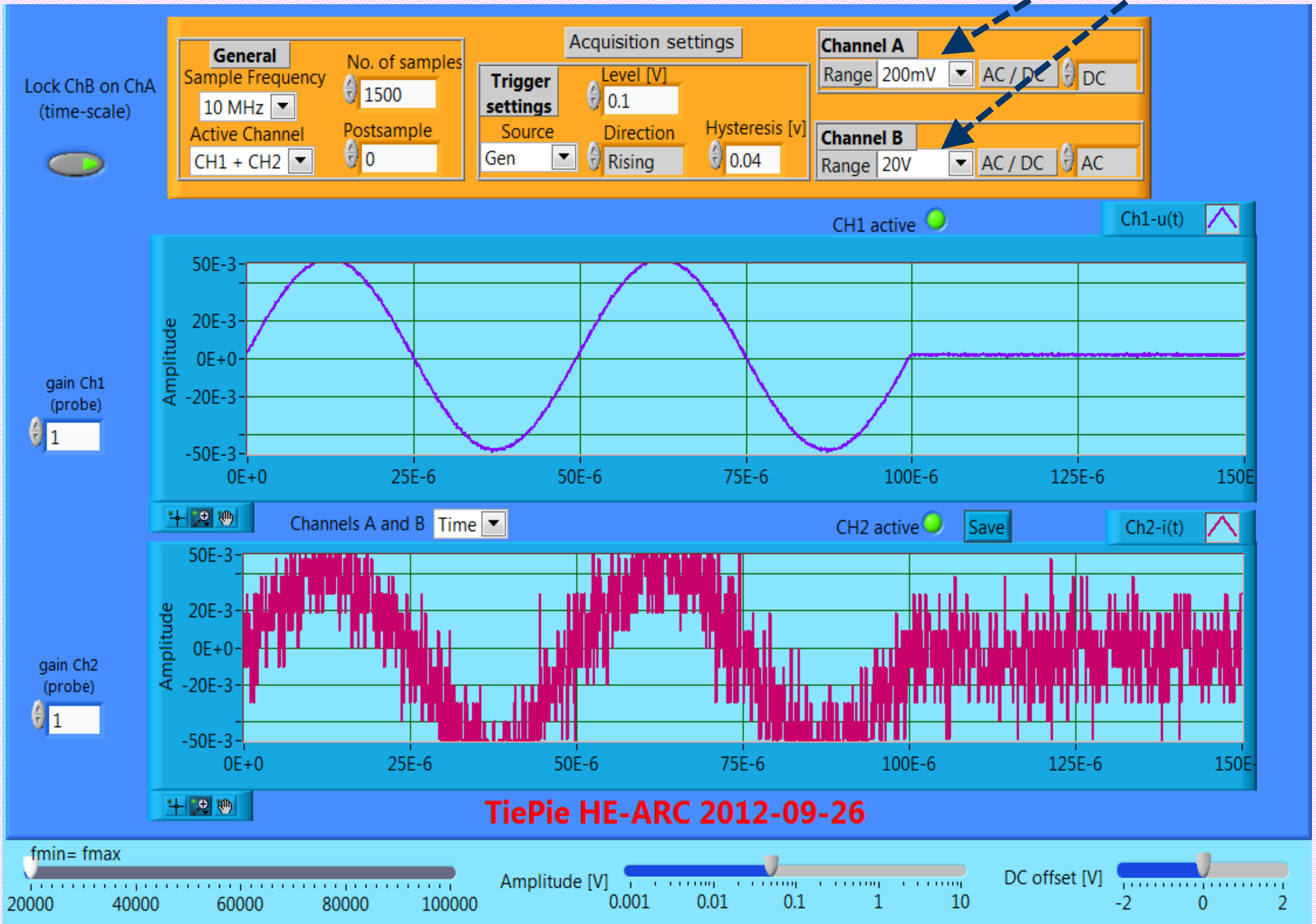
$x_d(nT)$ … 3 17 22 19 8 5 8 6 …

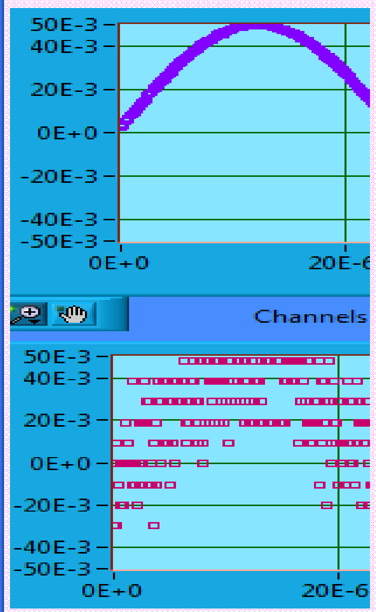
Quantization → Approximaton

→ Generation of noise

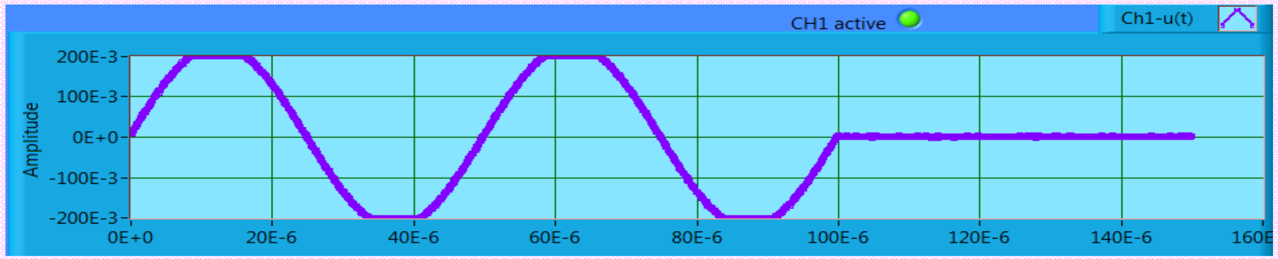
Acquisition de données : ***Quantification***

Estimation de quantification avec le TiePie HS3



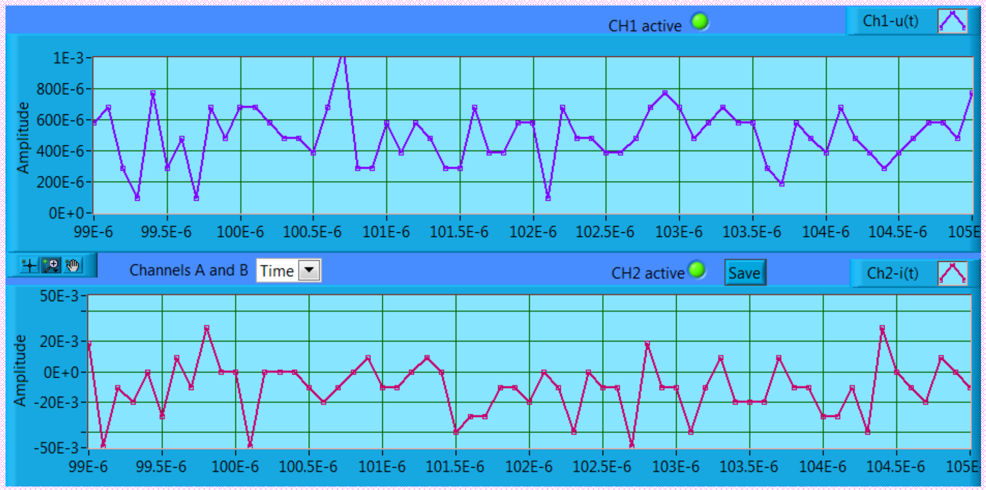


Acquisition de données : ***Quantification***



Range: 200mV
→ 400mVpp

Même « Range » mais entrées non connectées



Pas de quantification : 100µV

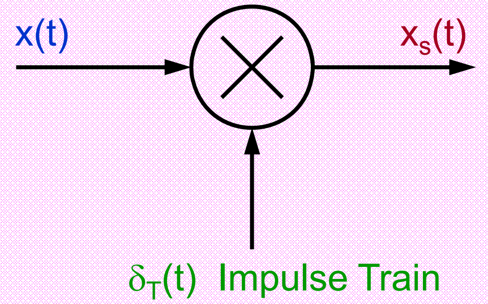
400mV / 100 µV = 4000 ≈ 2^{12} → ***12 bits***

Pas de quantification : 10mV

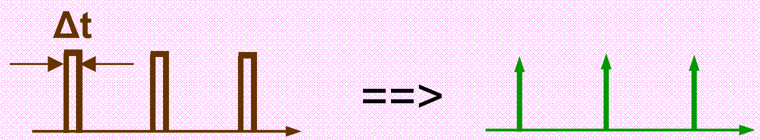
40V / 10 mV = 4000 ≈ 2^{12} → ***12 bits***

Acquisition de données : ***Echantillonnage***

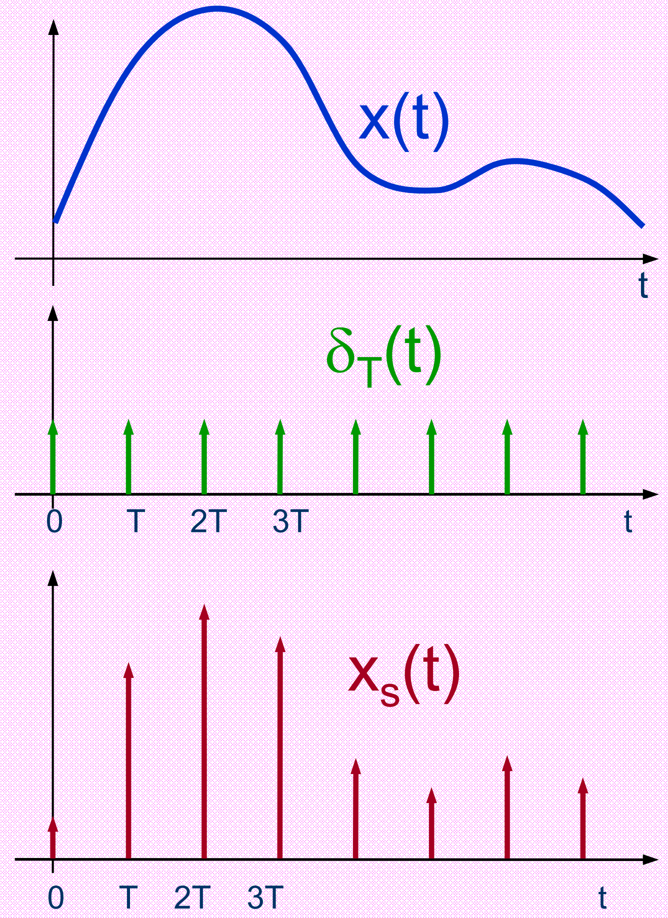
Sampling



In reality, the impulse train ***$\delta_T(t)$*** is a pulse train ***$p_T(t)$***

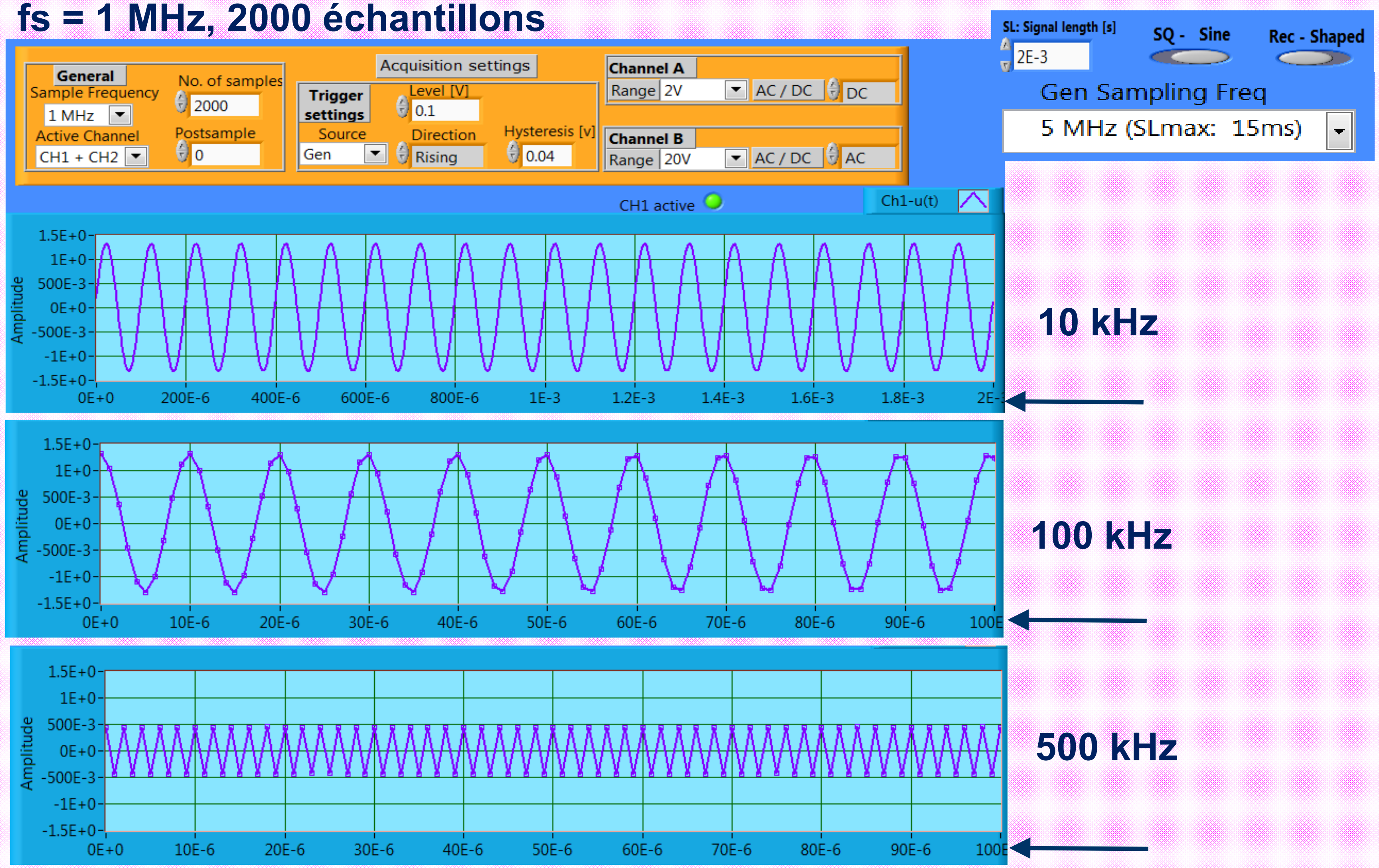


This approximation is valid if x(t) is ***almost*** *constant during a time interval* ***Δt***



Acquisition de données : ***Echantillonnage (1)***

fs = 1 MHz, 2000 échantillons



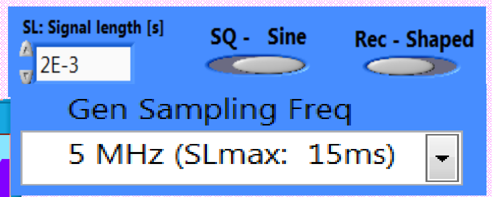
10 kHz

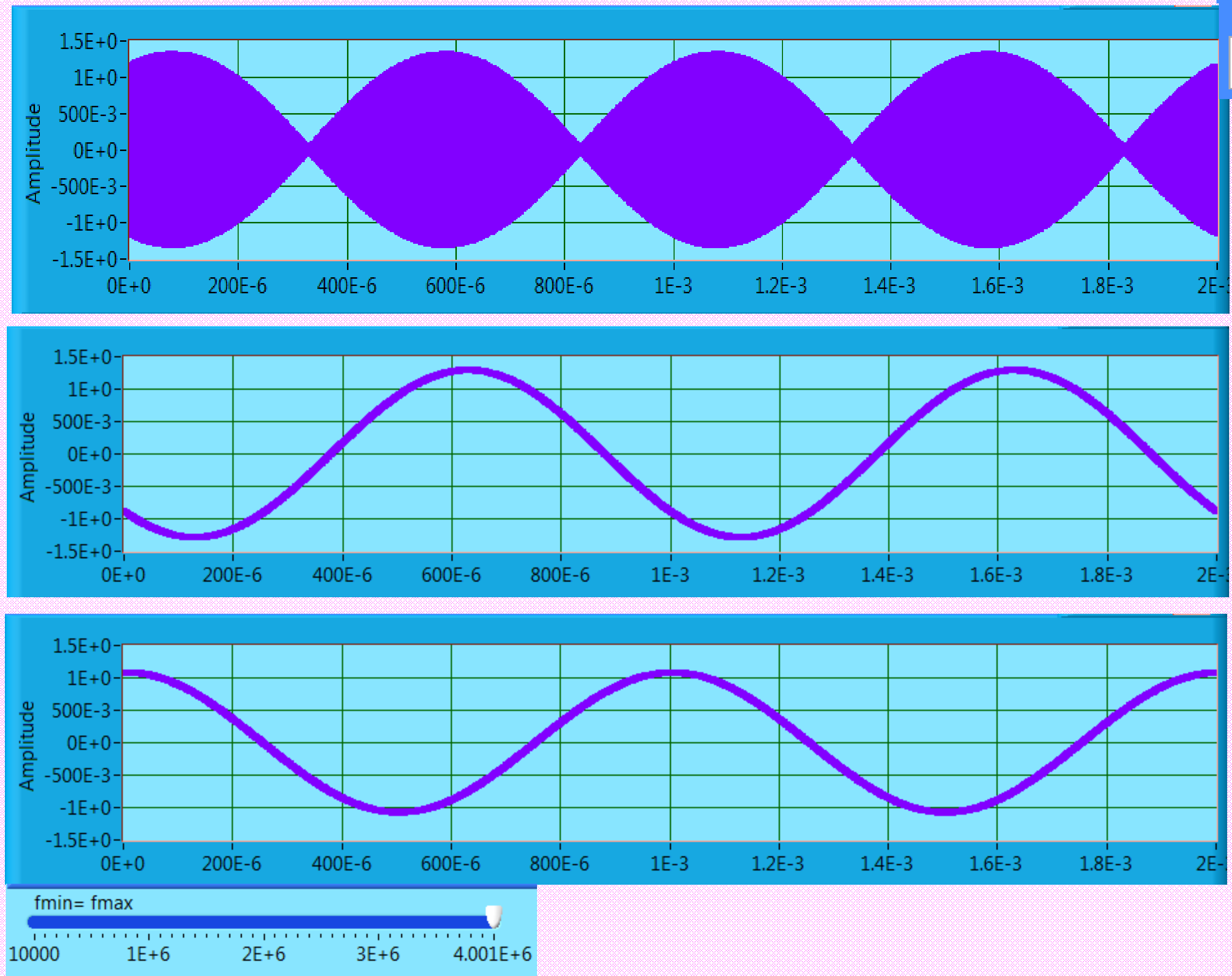
100 kHz

500 kHz

Acquisition de données : ***Sous-échantillonnage (1)***

fs = 1 MHz, 2000 échantillons



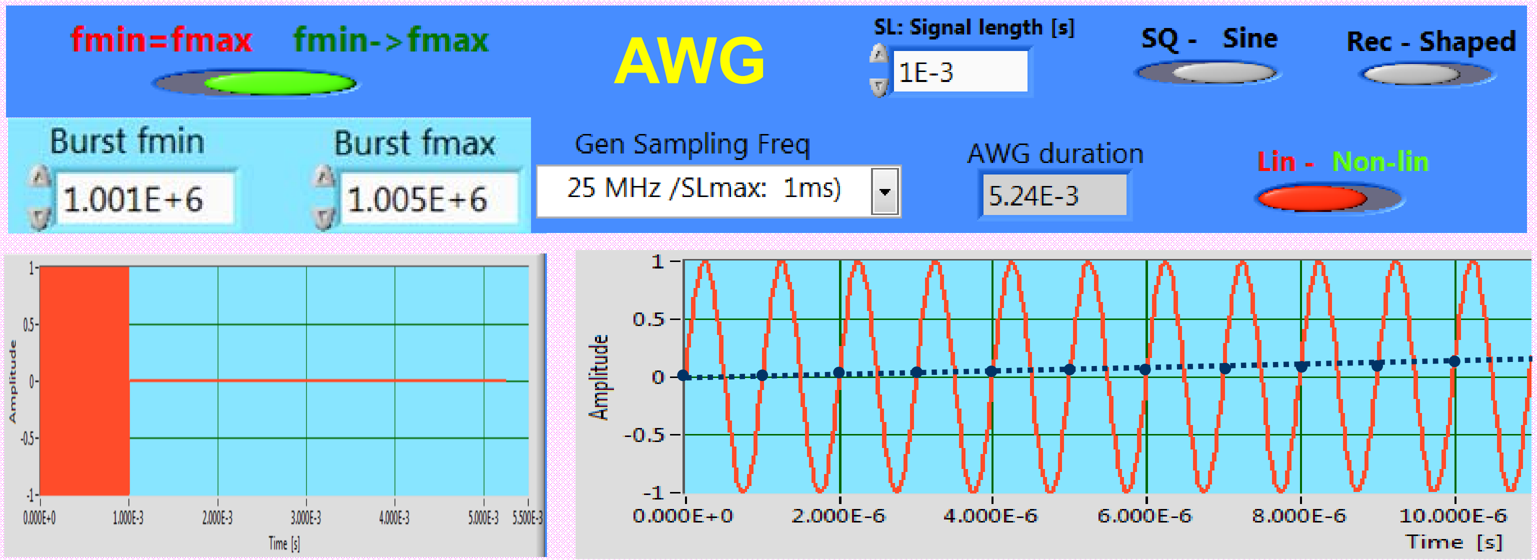


501 kHz

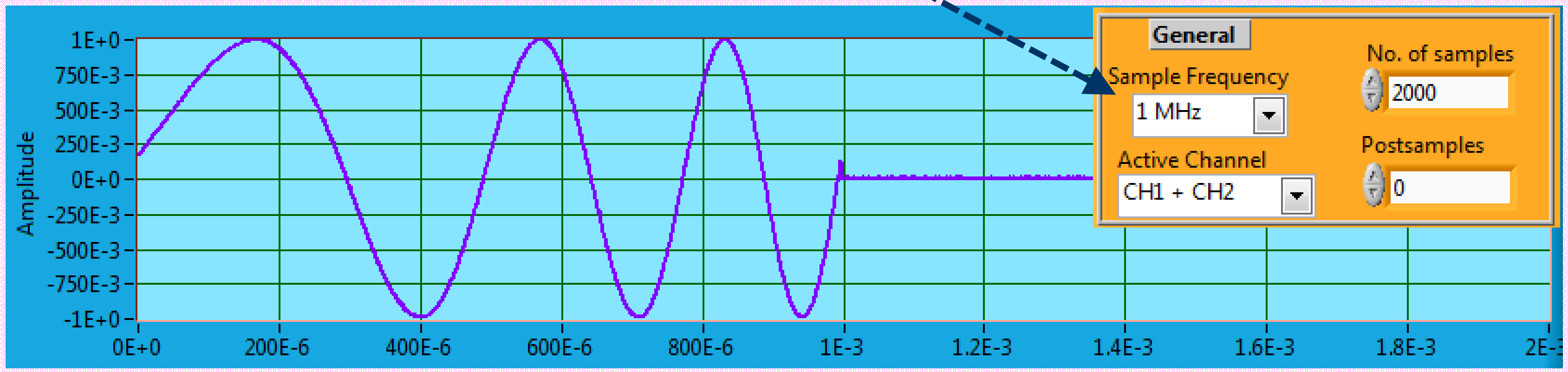
1001 kHz

4001 kHz

Acquisition de données : ***Sous-échantillonnage (2)***



1µs entre chaque échantillon



Acquisition de données : Problèmes

Acq-1

On doit numériser un signal de forme sinusoïdale dont la fréquence varie entre 10Hz et 1000 Hz. Si l'on considère qu'un minimum de 10 échantillons par période est nécessaire pour représenter et traiter le signal numérisé, quelle fréquence d'échantillonnage minimum faudra-t-il utiliser ?

Acq-2

Un signal issu d'un détecteur de marche de montre est de forme carrée asymétrique et d'une fréquence de 8Hz.

a) Quelle est la durée minimum d'acquisition nécessaire pour en déterminer la fréquence ?

b) Quelle fréquence d'échantillonnage minimum faut-il utiliser si l'on désire obtenir une précision relative de 10^{-5} (10ppm) ?

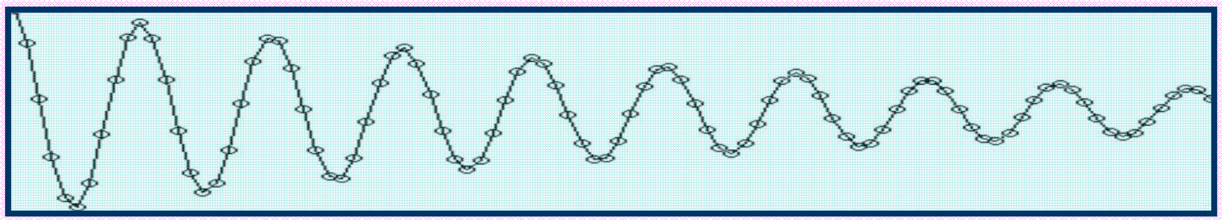
c) Si l'on considère 100 périodes de notre signal, quelle sera alors la nouvelle fréquence d'échantillonnage (même précision relative) ?

Acq-3

Un signal x(t) peut-être représenté comme suit :

$$x(t) = A \cdot e^{\frac{-t}{\tau}} \cdot \sin\left(2 \cdot \pi \cdot 60 \cdot t - \frac{\pi}{4}\right)$$

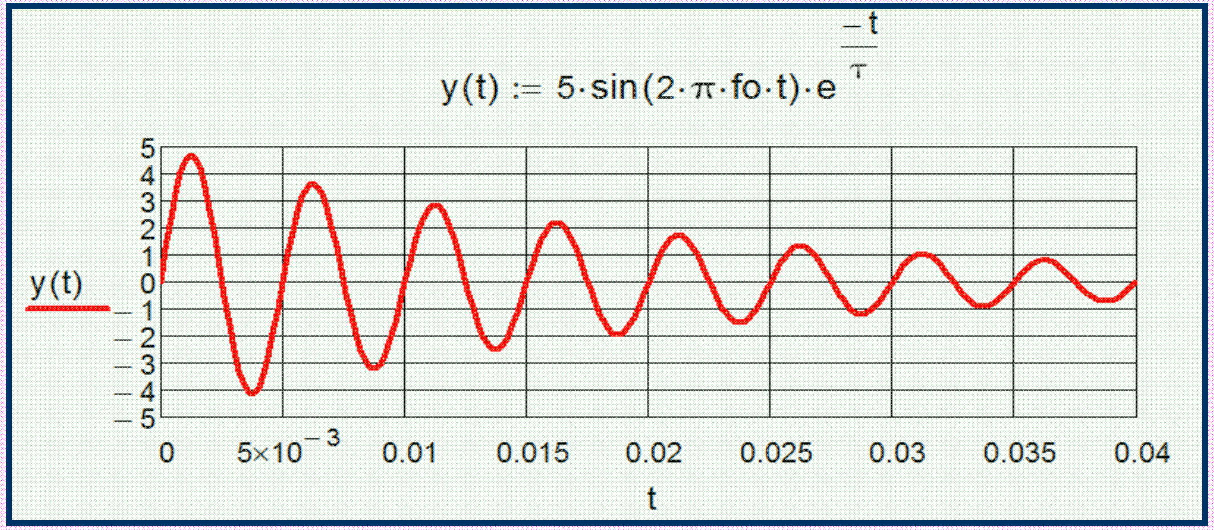
Après échantillonnage, le signal est le suivant :



Quelle fréquence d'échantillonnage approximative a été utilisée ?

Acquisition de données : Problèmes

Acq-4



Un convertisseur A/D de 12 bits couvre la plage de ±10V.

a) Déterminer le pas de quantification.

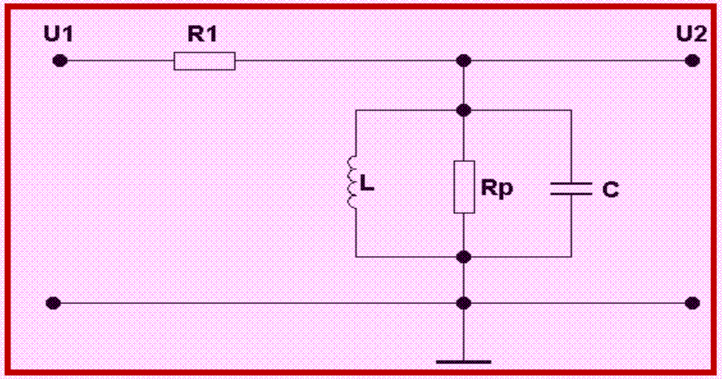
b) Combien de périodes du signal pourra-t-on acquérir tout en garantissant une erreur relative sur l'amplitude maximum de 20% ?

c) Si l'on désire doubler ce nombre de périodes tout en acceptant d'en écrêter les premières, quel gain faut-il mettre à un préamplificateur placé avant le convertisseur et combien de périodes seront ainsi écrêtées ?

d) Combien de bits faut-il choisir si l'on veut un pas de quantification plus petit que 500µV ?

B1: Test de la fonction de transfert d'un filtre du 2ème ordre

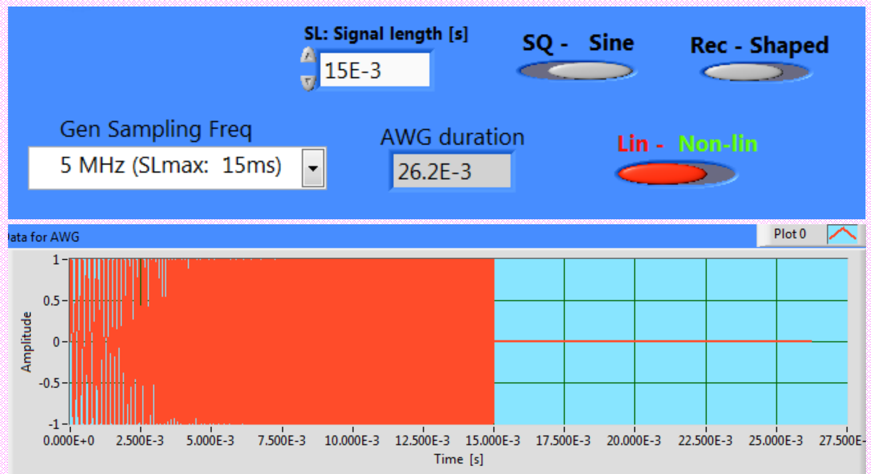
a) Vérifier le comportement en passe-bande de ce filtre, déterminer Rp



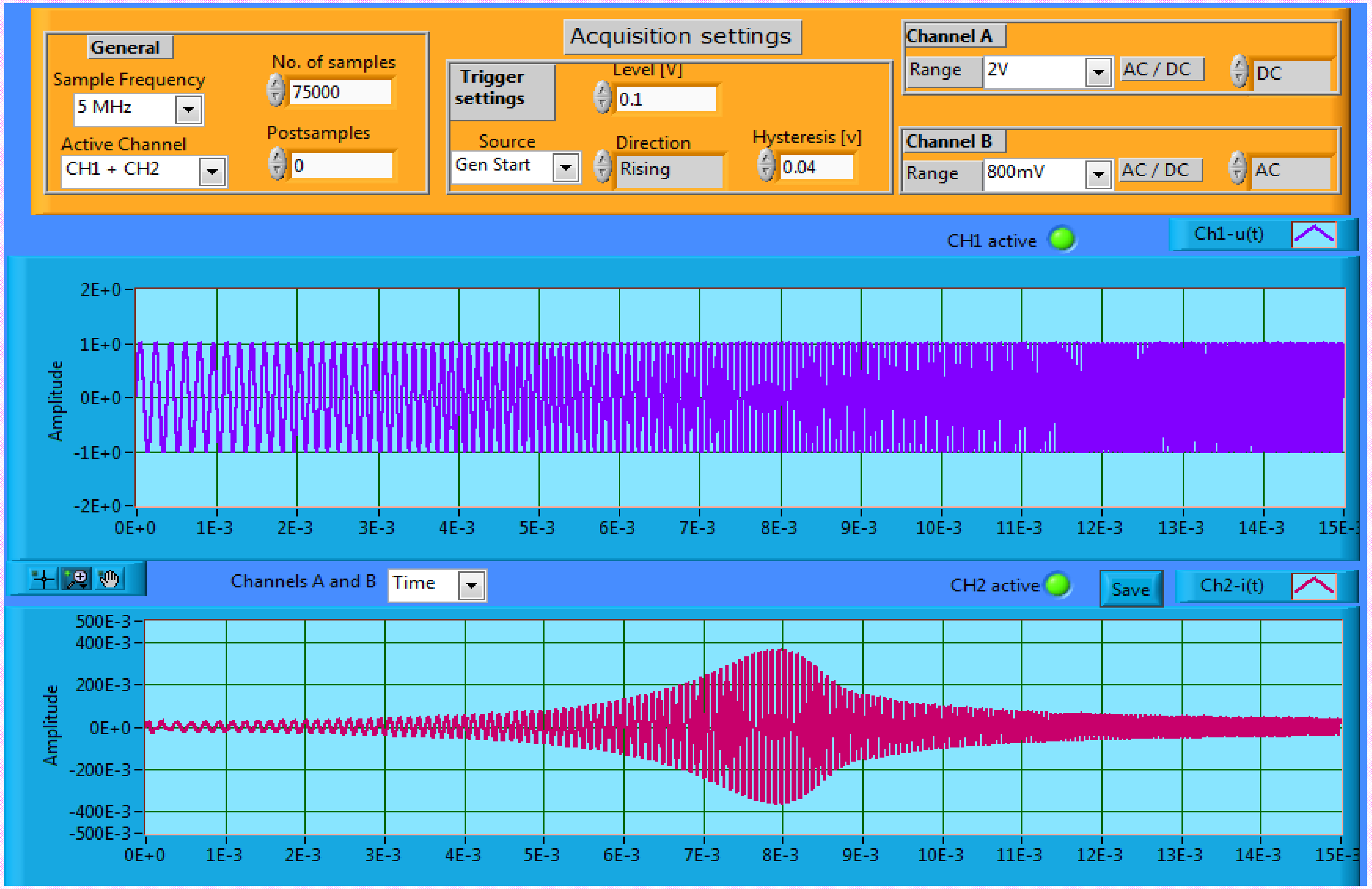
R1: 220 kΩ
L: 100 mH
C: 1.5 nF
Rp: Pertes de L
$f_r = 1 / [2\cdot\pi\cdot (L\cdot C)^{0.5}]$

$$\frac{1}{2\cdot\pi\sqrt{L\cdot C}} = 12.995\times10^{3}$$

b) Utiliser le balayage en fréquence pour démontrer l'effet passe-bande filtre en paramétrant LabView comme suit:

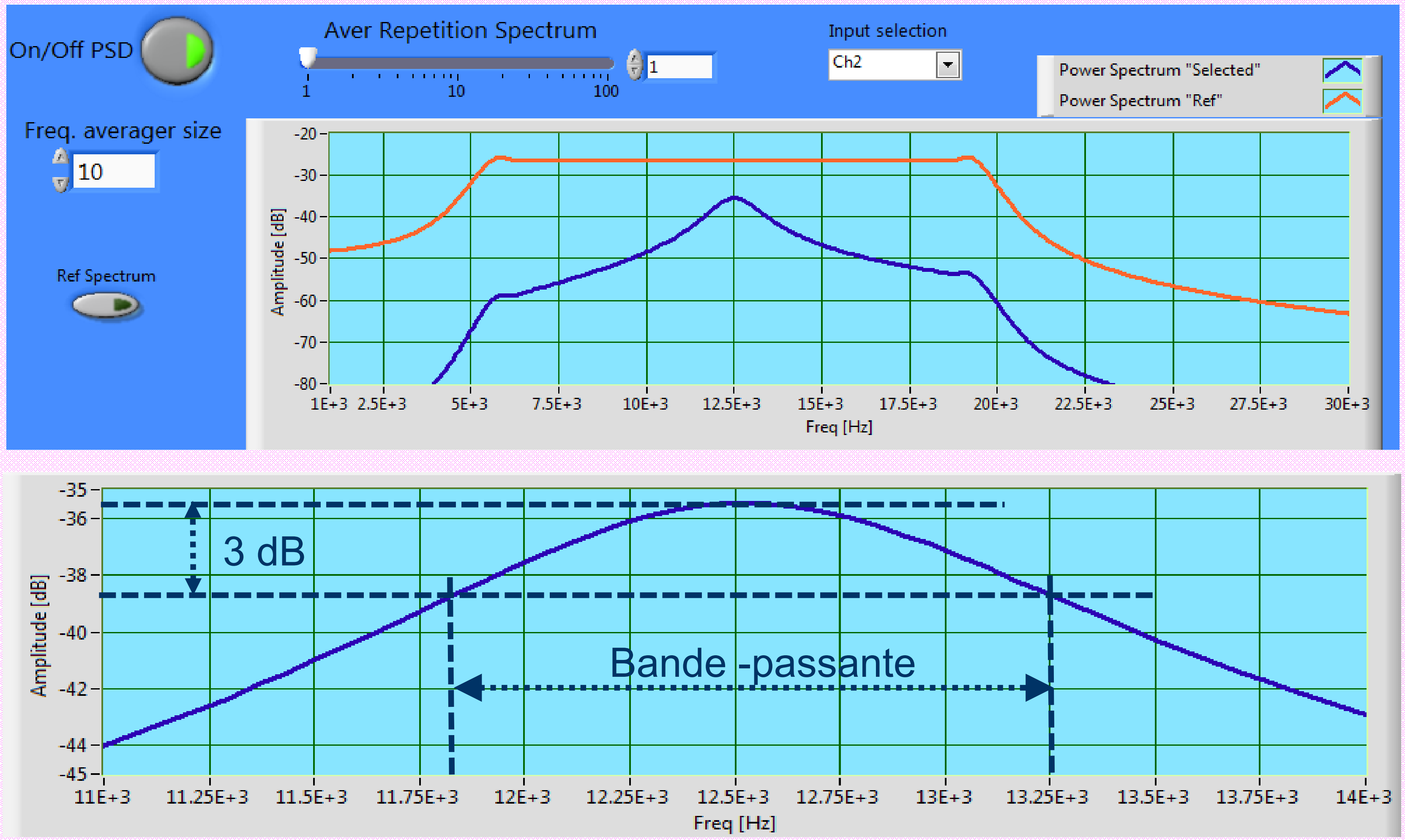


B1: Test de la fonction de transfert d'un filtre du 2ème ordre



B1: Caractéristique spectrale d'un filtre passe-bande R-L-C

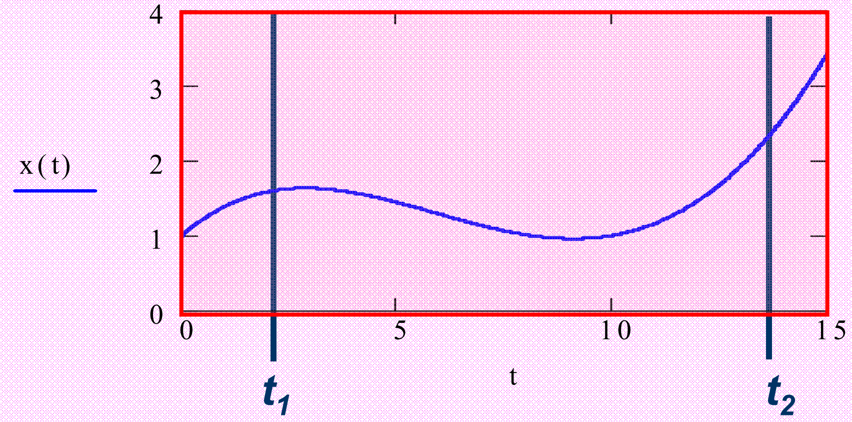
Utilisation de l'environnement « Power Spectrum »



SERIES *(review)*: *Definition and Concept*

Within a given time interval, a signal x(t) can be approximated by a linear combination of *appropriately preselected* N orthogonal functions, also called basis functions as:

$$\hat{x}(t) = \sum_{n=0}^{N} a_n \cdot \Phi_n(t)$$



Validity interval: $t_1 \le t \le t_2$

With $\Phi(t)_i$: ith orthogonal function
a_i : ith coefficient

Objective:

To represent x(t) between t_1 and t_2 with a minimum of coefficients!

Orthogonality:

$$\int_{t_1}^{t_2} \Phi_n(t) \cdot \Phi_k(t)\, dt = 0 \quad \text{for all } n \ne k$$

Frequently used criterion:

$$\varepsilon = \int_{t_1}^{t_2} (\hat{x}(t) - x(t))^2\, dt$$

FOURIER SERIES *(review)*

Definition:

$$f(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} \left(a_n \cdot \cos(n \cdot \omega_0 \cdot t) + b_n \cdot \sin(n \cdot \omega_0 \cdot t) \right)$$

With:

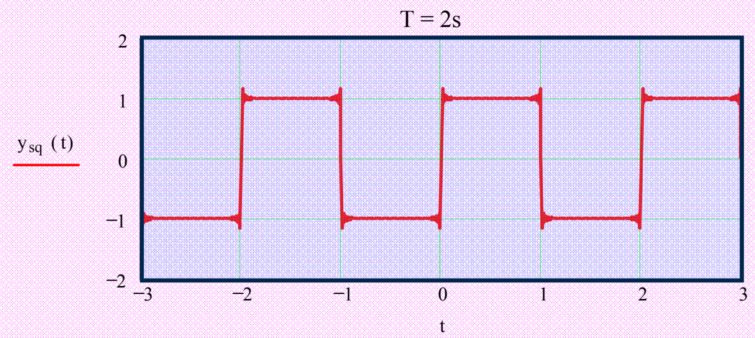
$$a_n = \frac{2}{T} \cdot \int_{t_1}^{t_1+T} f(t) \cdot \cos(n \cdot \omega_0 \cdot t)\, dt \qquad b_n = \frac{2}{T} \cdot \int_{t_1}^{t_1+T} f(t) \cdot \sin(n \cdot \omega_0 \cdot t)\, dt$$

For: n: 0, 1, 2, 3… n: 1, 2, 3…

In general, the series represents f(t) over the time interval t_1 to t_1+T, and nothing is specified about f(t) outside this interval. ***However, if f(t) is periodic with a period T, then the Fourier series representation will be valid for all t.***

Example: Fourier Series of a symetric square-wave

$$y_{sq}(t) := \frac{4}{\pi} \cdot \sum_{n=1}^{N} \frac{1}{2n-1} \cdot \sin\left[2 \cdot \pi \cdot \frac{(2 \cdot n - 1) \cdot t}{T} \right]$$



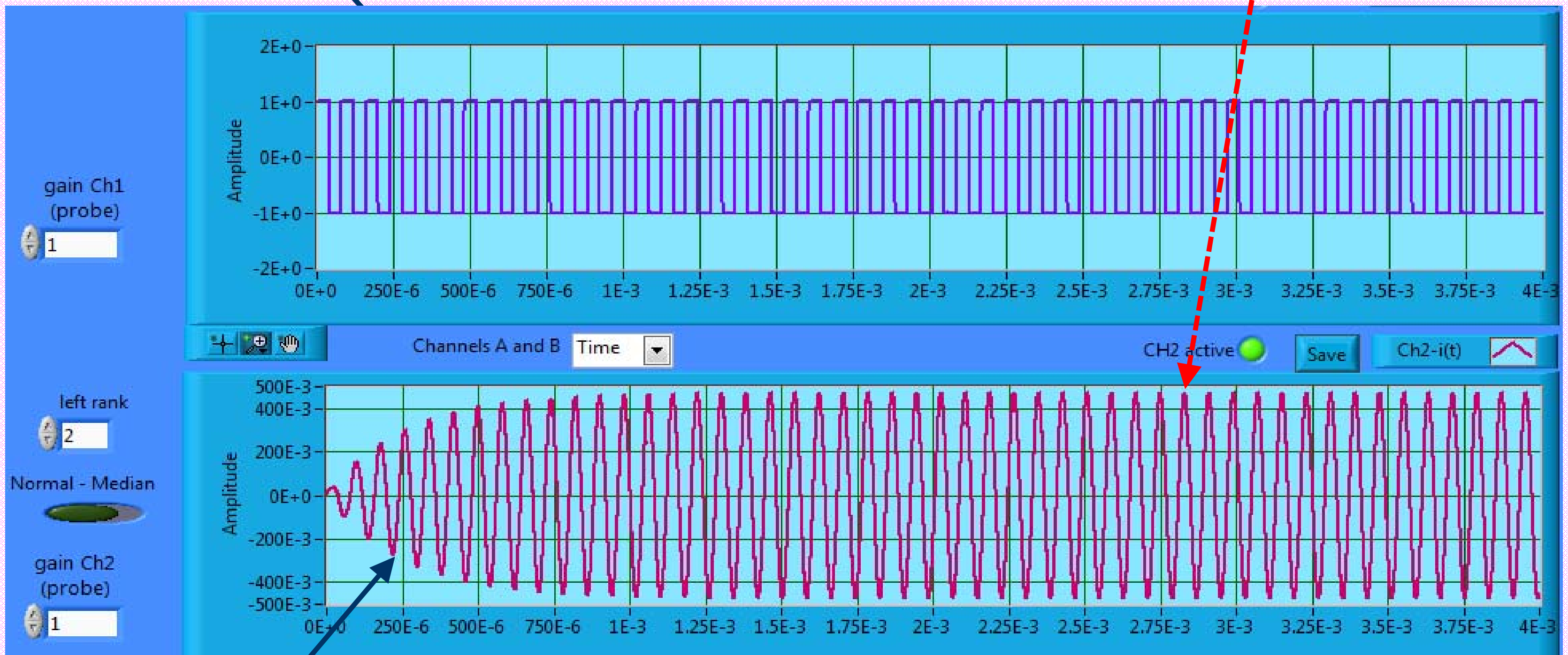
Application d'un filtre passe-bande (1)

Séries de Fourier (1)

Signal carré de 13 kHz
(13 kHz, 39 kHz, 65kHz…)

Passe-bande R-L-C
$f_r \approx 13$ kHz, Bw ≈ 1.5 kHz

Sinus?



Temps de montée du filtre qui est environ l'inverse de sa bande-passante !

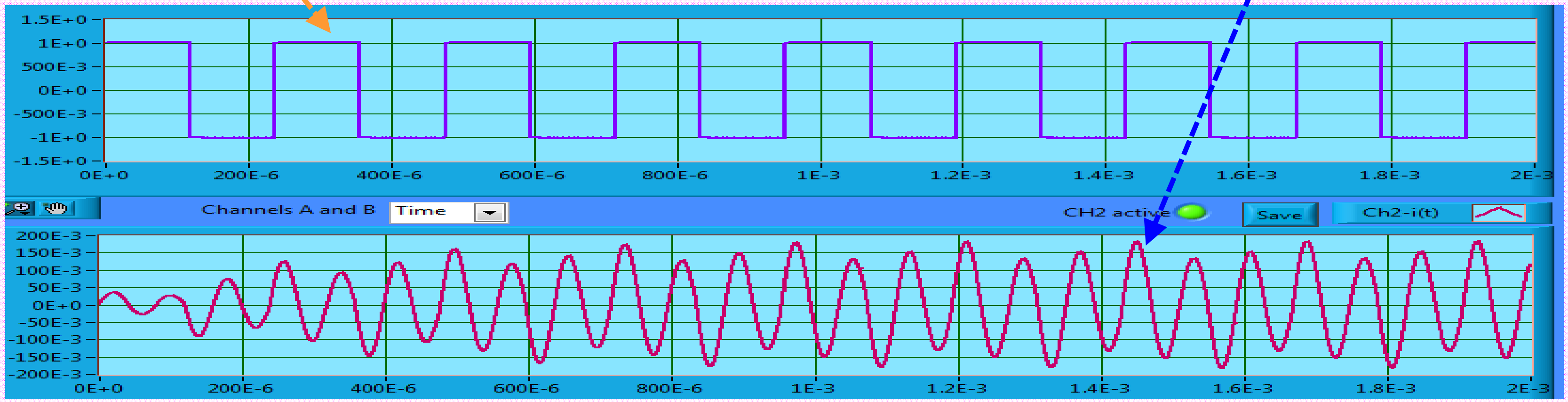
Application d'un filtre passe-bande (1a)

Séries de Fourier (2)

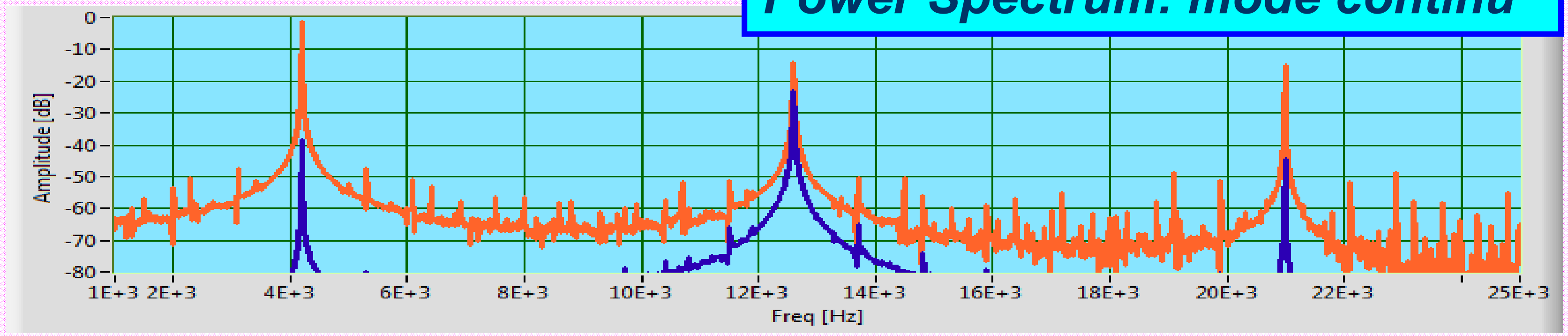
Signal carré de 4.33 kHz
(4.33 kHz, 13 kHz, 21.7kHz…)

Passe-bande R-L-C
fr ≈ 13 kHz, Bw ≈ 1.5 kHz

Sinus + harmonique ?

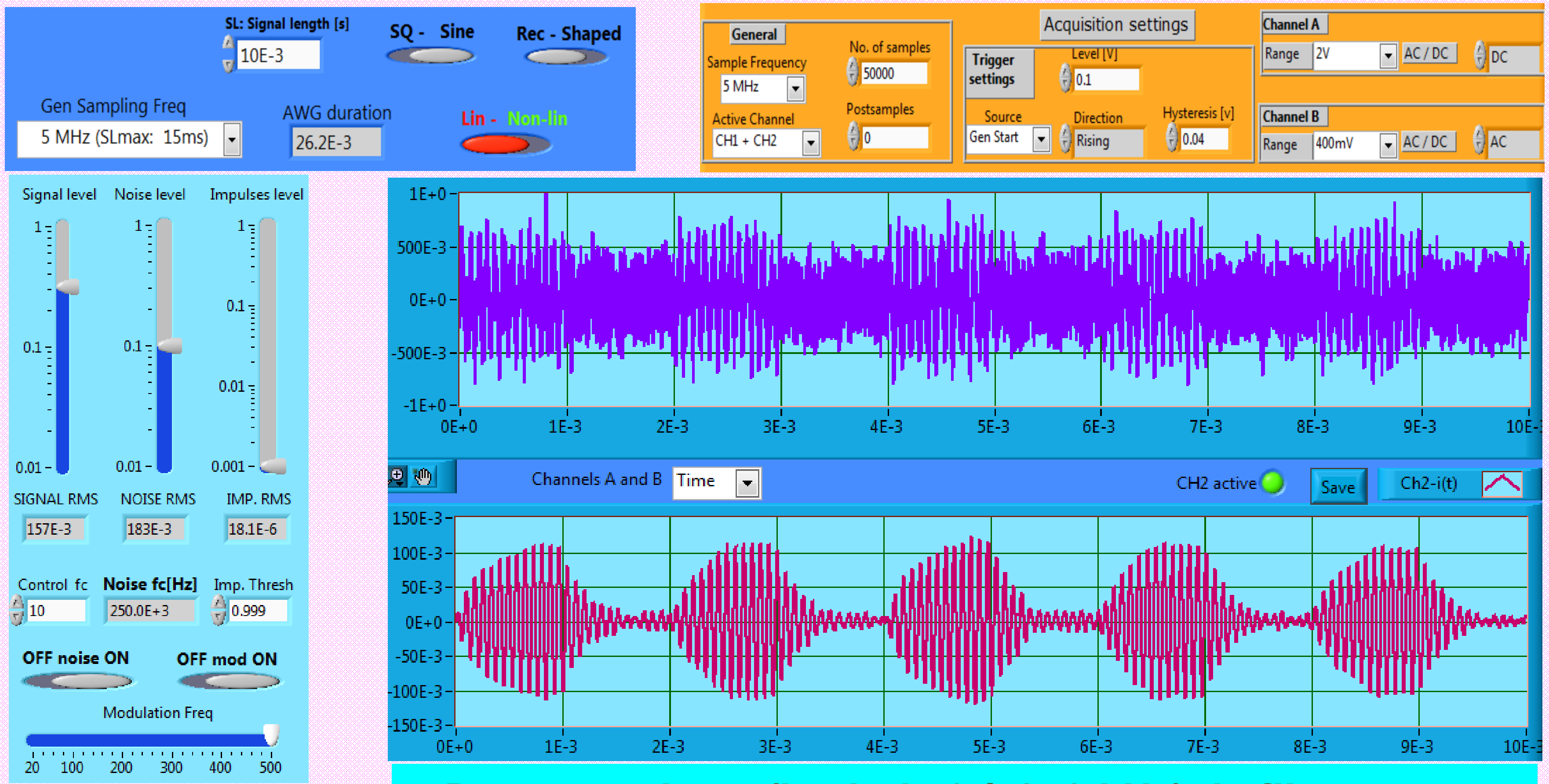


Power Spectrum: mode continu



Application d'un filtre passe-bande (2)

Réduction du bruit additionné à un signal modulé de type "ON-OFF keying"

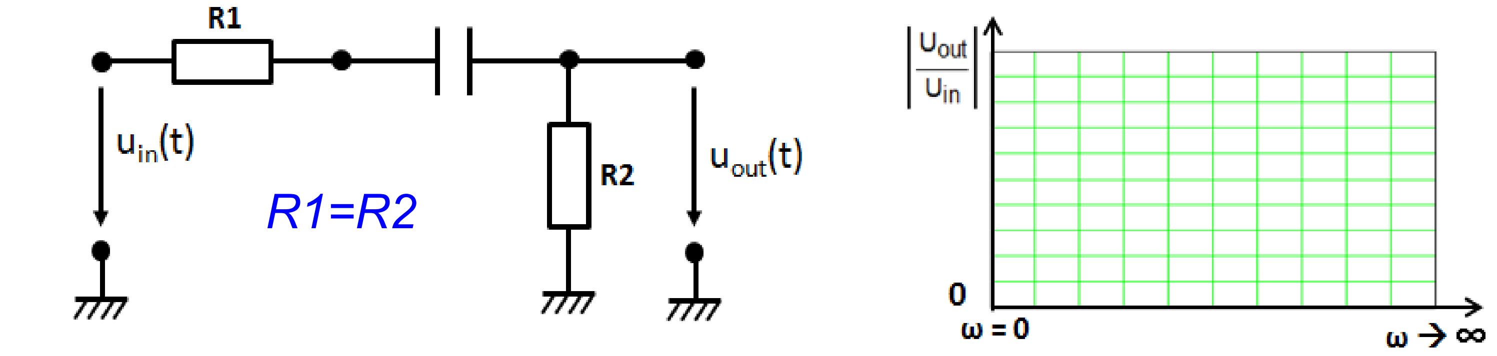


13 kHz, fmin=fmax

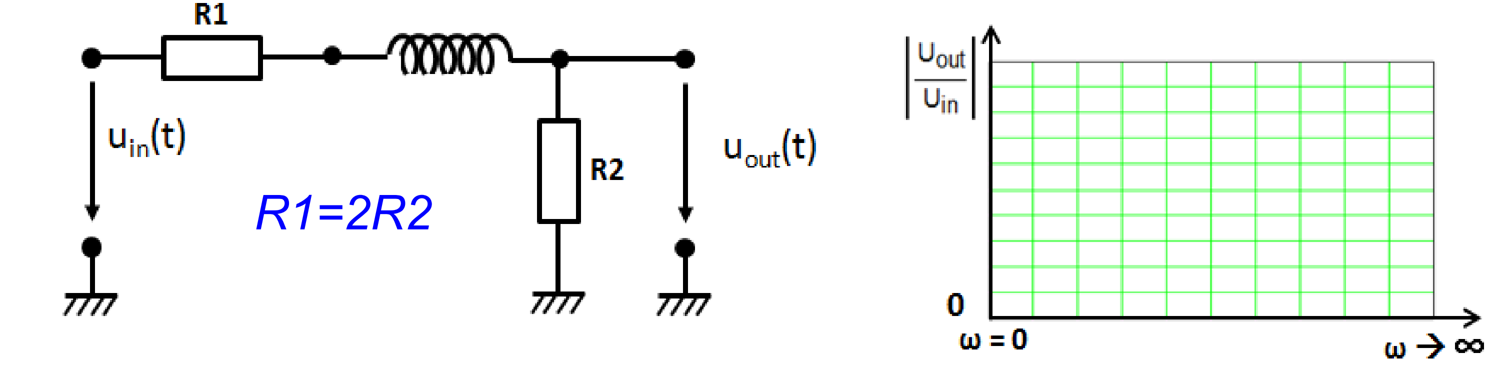
Dans un tel cas (bruit de 0 à 250 kHz), le filtre passe-bande réduit fortement le bruit

Détermination "rapide" du type de filtre (1)

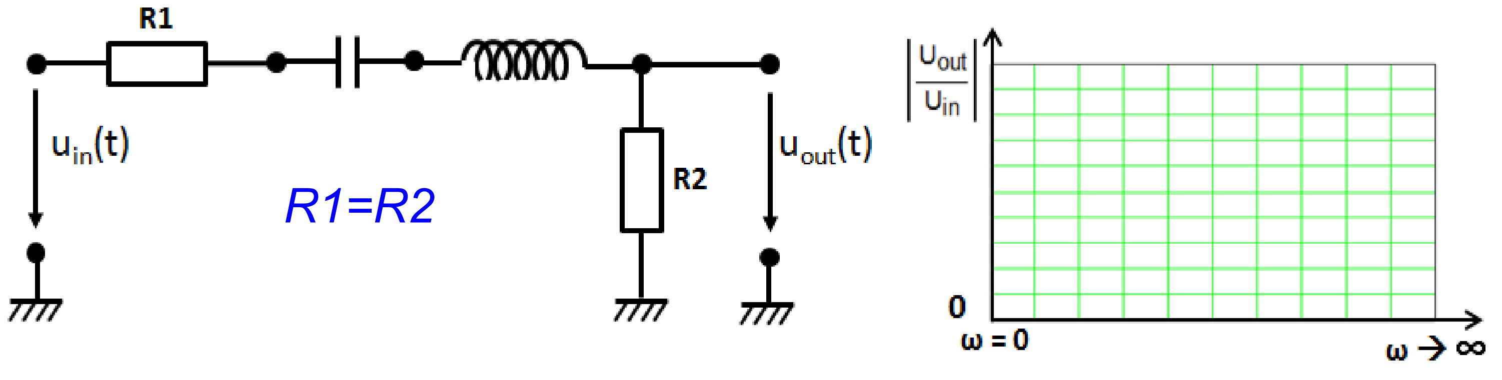
a)



b)

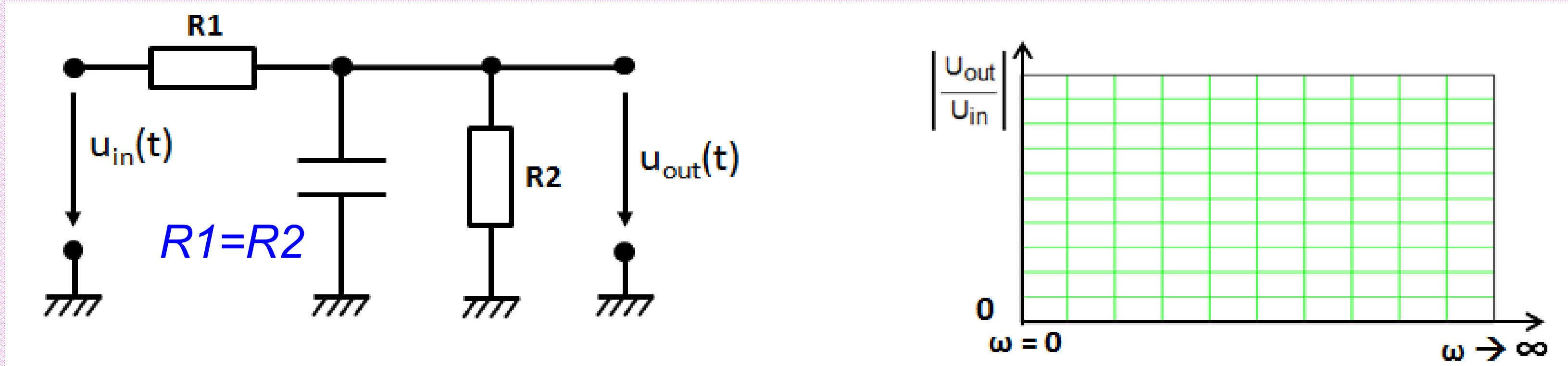


c)

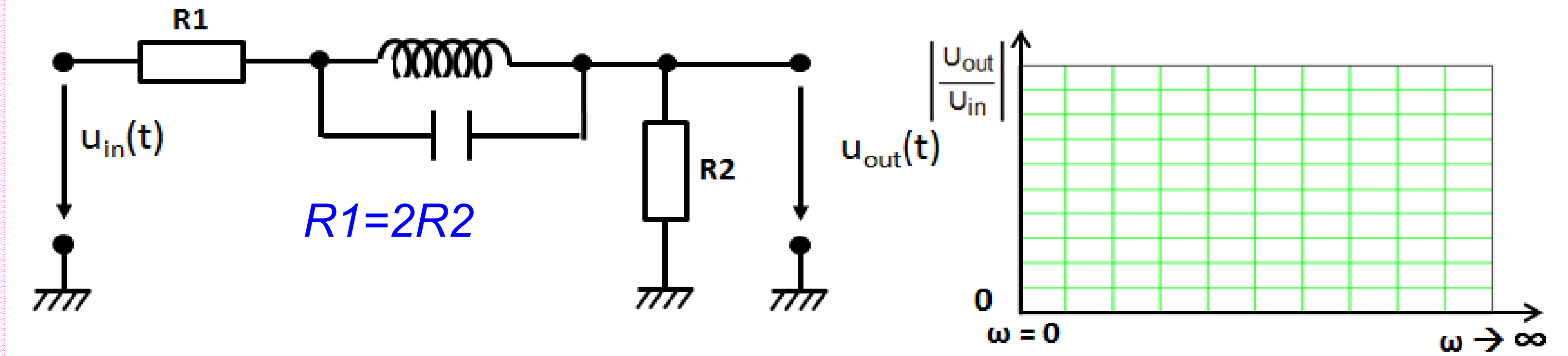


Détermination "rapide" du type de filtre (2)

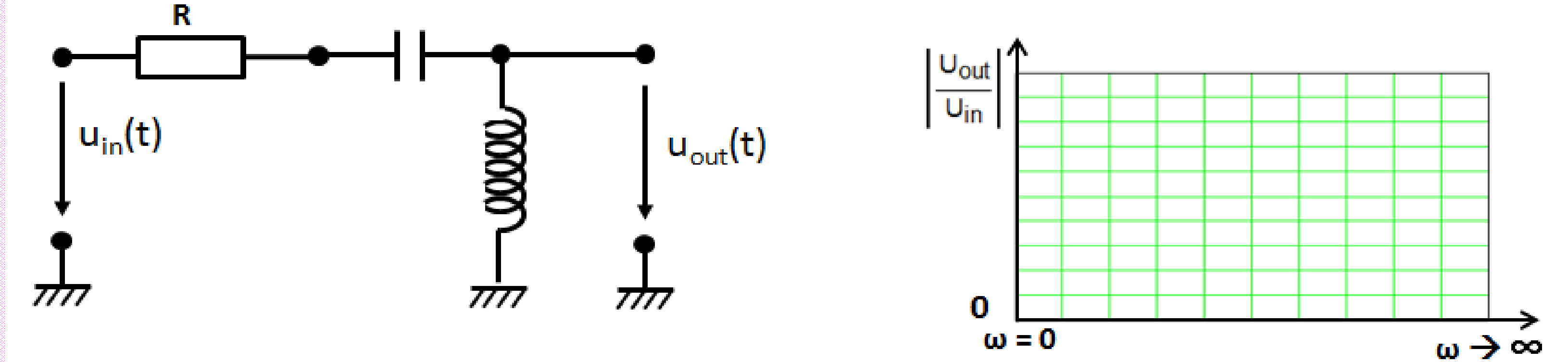
d)



e)

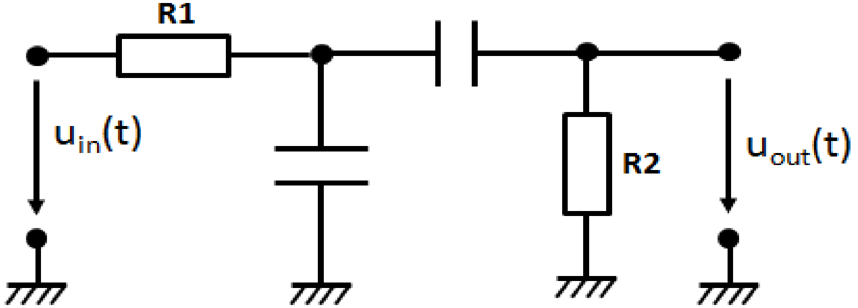


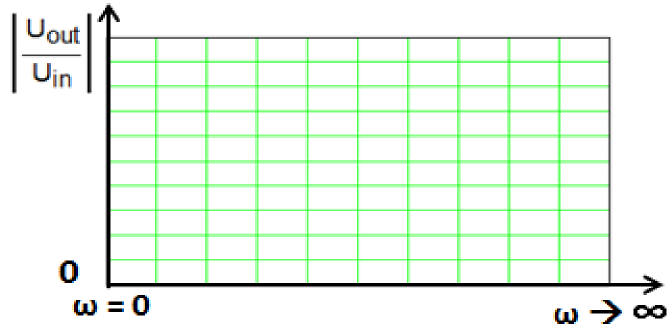
f)



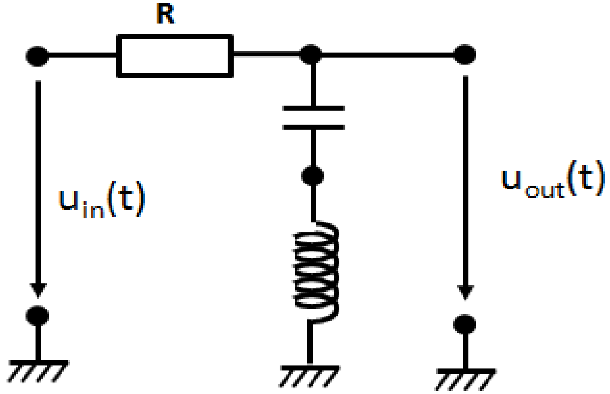
Détermination "rapide" du type de filtre (3)

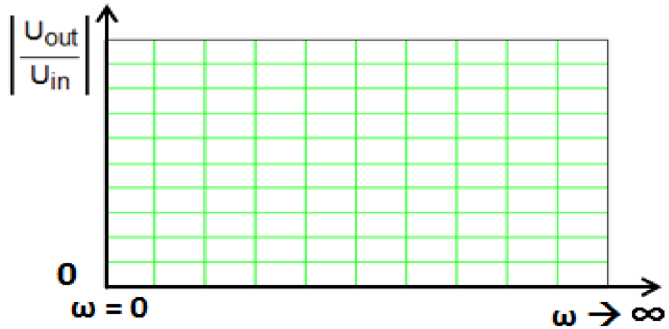
g)



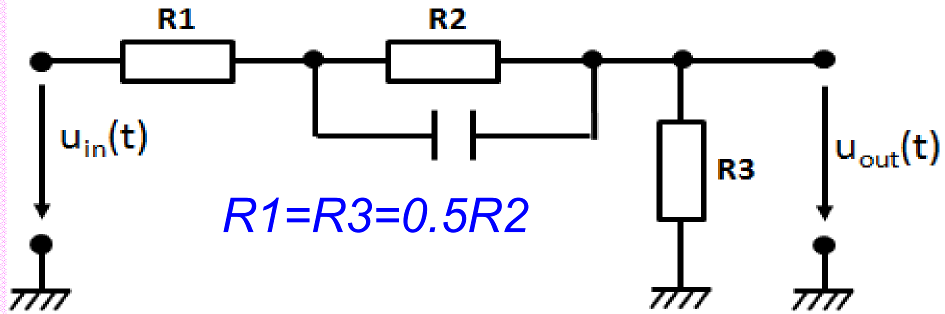


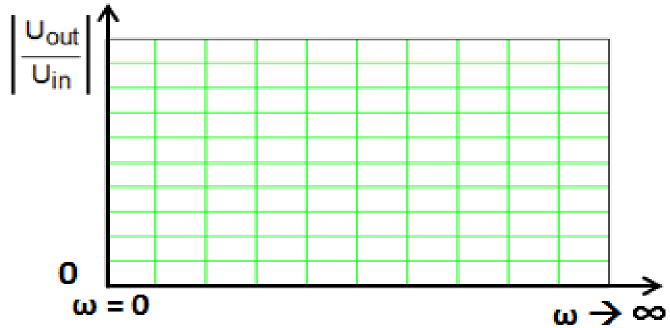
h)





i)





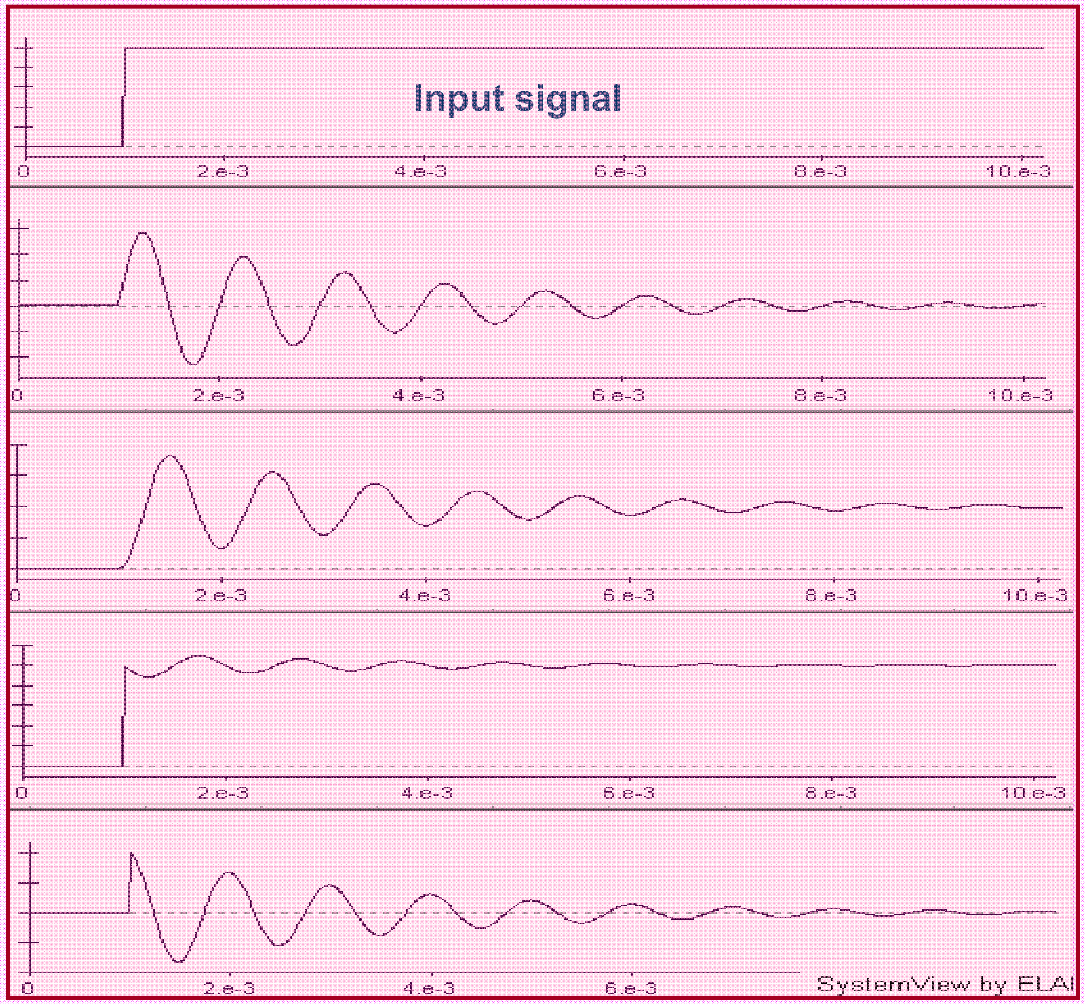
Réponses à un saut unitaire de différents filtres du 2ème ordre

Filtre passe-bande

Filtre passe-bas

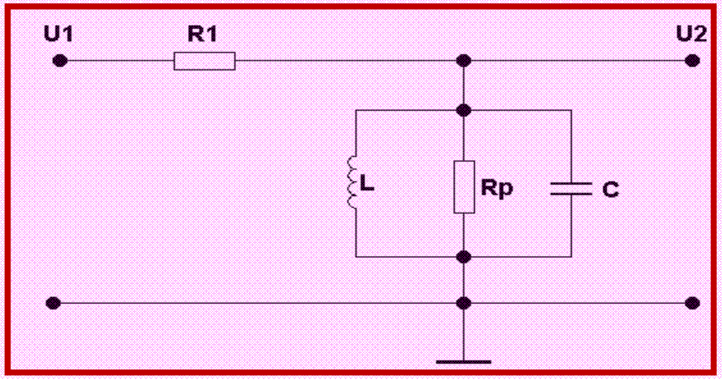
Filtre réjecteur

Filtre passe-haut



Test de la réponse transitoire d'un filtre du 2ème ordre (1)

Calculer et mesurer la réponse à un saut unitaire de ce filtre du 2ème ordre.



R1: 220 kΩ, Rp: 110 kΩ
L: 100 mH
C: 1.5 nF
Rp: Pertes de L
f_r = 1 / [2·π· (L·C)$^{0.5}$]

$$Req = \frac{R1 \cdot Rp}{R1 + Rp}$$

$$\frac{U_2(s)}{U_1(s)} = H(s) = \frac{\dfrac{S \cdot L \cdot \frac{1}{S \cdot C}}{S \cdot L + \frac{1}{S \cdot C}} \cdot \dfrac{Rp}{R1 + Rp}}{Req + \dfrac{S \cdot L \cdot \frac{1}{S \cdot C}}{S \cdot L + \frac{1}{S \cdot C}}} = \frac{\dfrac{s}{C \cdot Req} \cdot \dfrac{Rp}{R1 + Rp}}{s^2 + \dfrac{s}{C \cdot Req} + \dfrac{1}{L \cdot C}}$$

$$\rightarrow U_2(s) = \frac{1}{s} \cdot H(s) = \frac{\dfrac{1}{C \cdot Req} \cdot \dfrac{Rp}{R1 + Rp}}{s^2 + \dfrac{s}{C \cdot Req} + \dfrac{1}{L \cdot C}}$$

Saut unitaire

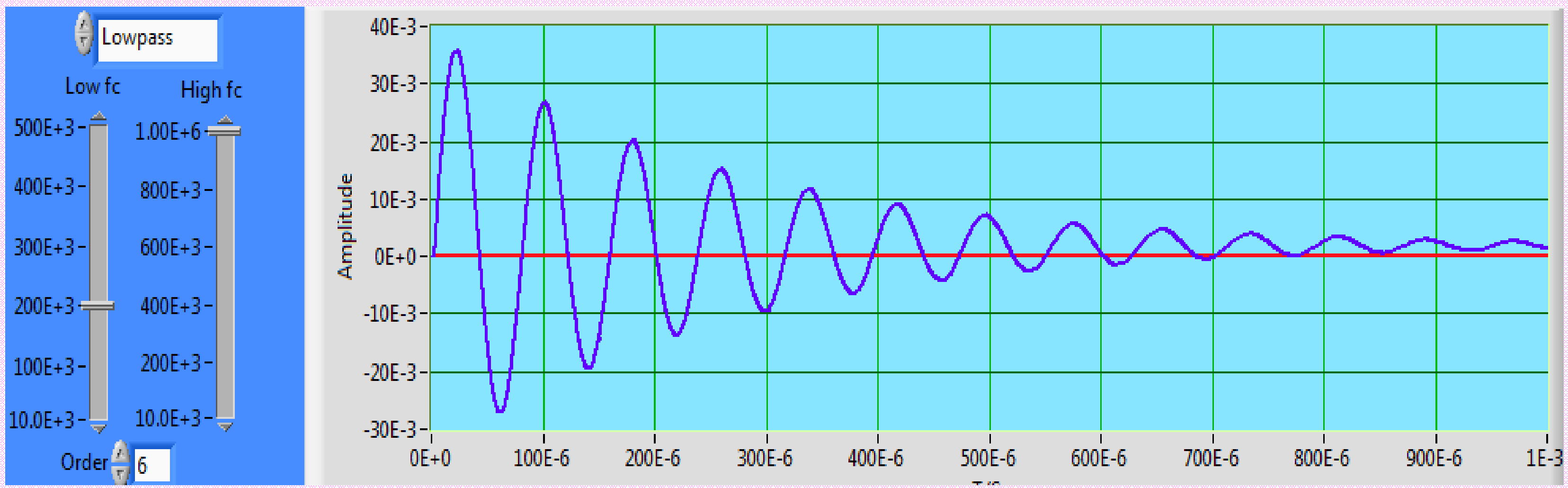
$$\Longrightarrow \quad u_2(t) = \frac{Rp}{R1 + Rp} \cdot k \cdot e^{-a \cdot t} \cdot \sin(\omega_0 \cdot t)$$

Test de la réponse transitoire d'un filtre du 2ème ordre (2)

$$\rightarrow \quad k = \frac{1}{\sqrt{\frac{C \cdot Req^2}{L} - \frac{1}{4}}} \qquad a = \frac{1}{2 \cdot C \cdot Req} \qquad \tau = 2 \cdot C \cdot Req \qquad \omega o = \sqrt{\frac{1}{L \cdot C} - \frac{1}{4 \cdot C^2 \cdot Req^2}}$$

$$Req = 73.3 \times 10^3 \qquad \frac{110}{110 + 220} \cdot k = 37.2 \times 10^{-3} \qquad 2 \cdot C \cdot Req = 220 \times 10^{-6} \qquad f_o = 13 \times 10^3$$

Réponse à un saut unitaire :



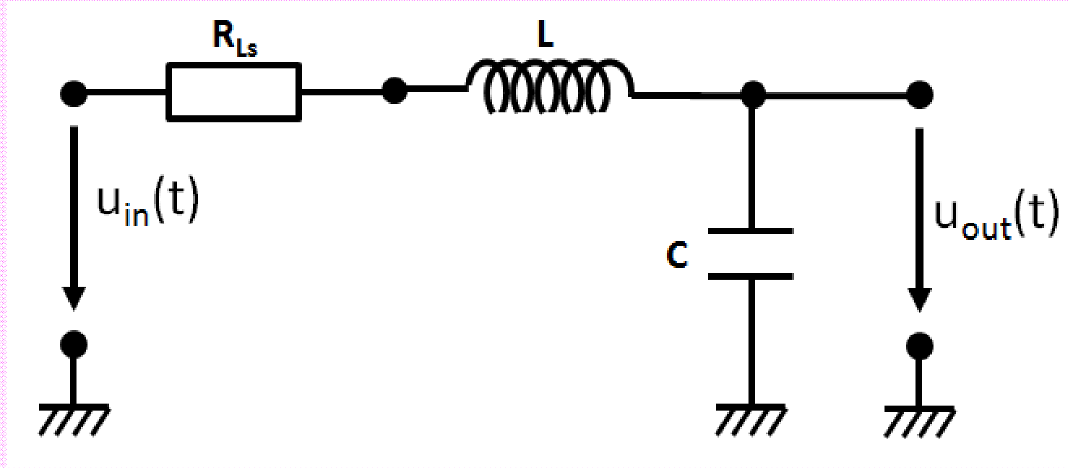
Test de la réponse transitoire d'un filtre du 2ème ordre (3)

De quel type de filtre s'agit-il ?

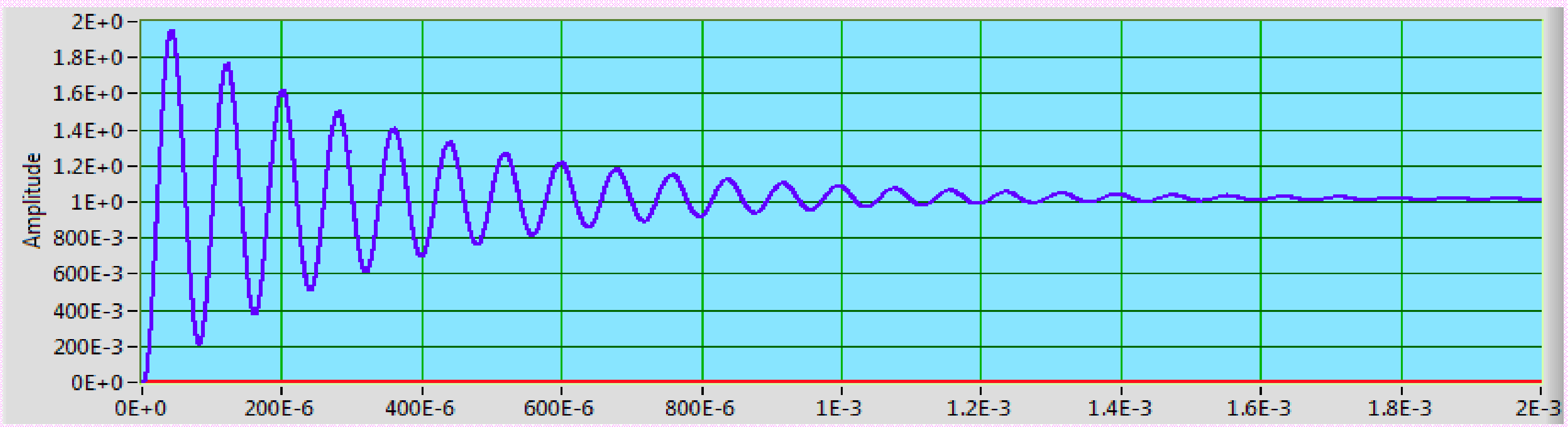
Déterminer pratiquement sa réponse à un saut unitaire

L=100mH C=1.5nF

R_{Ls}: résistance série de perte de L

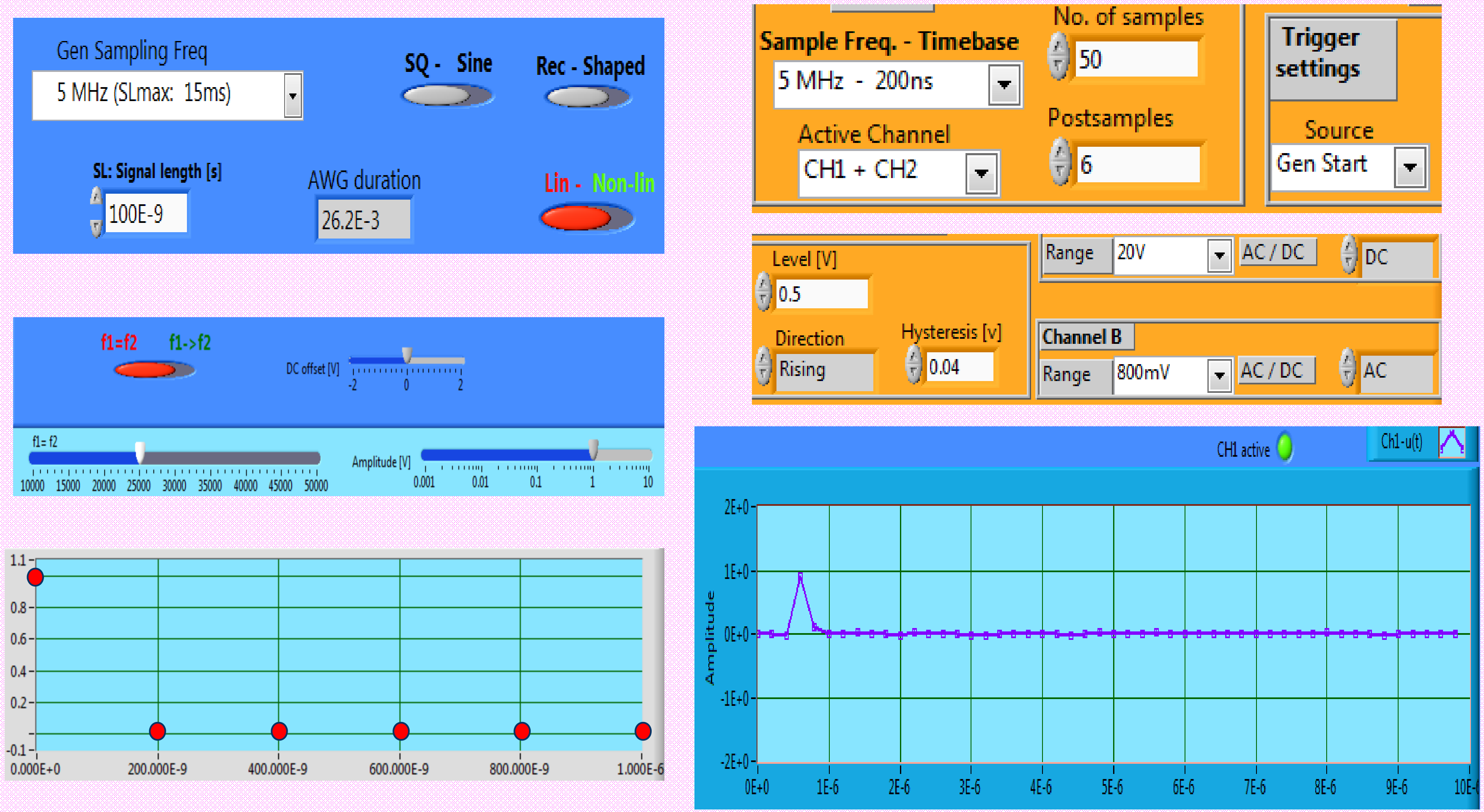


Réponse à un saut unitaire :

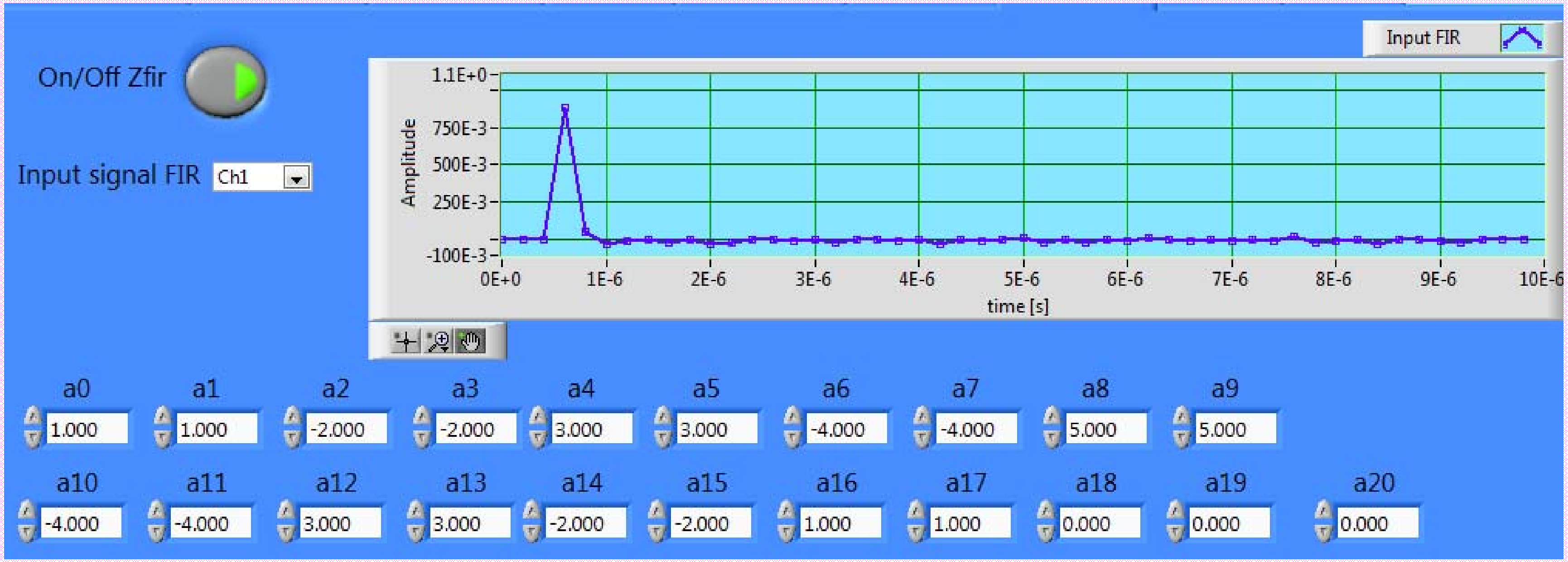


Filtres numériques par la transformation en Z : FIR (1)

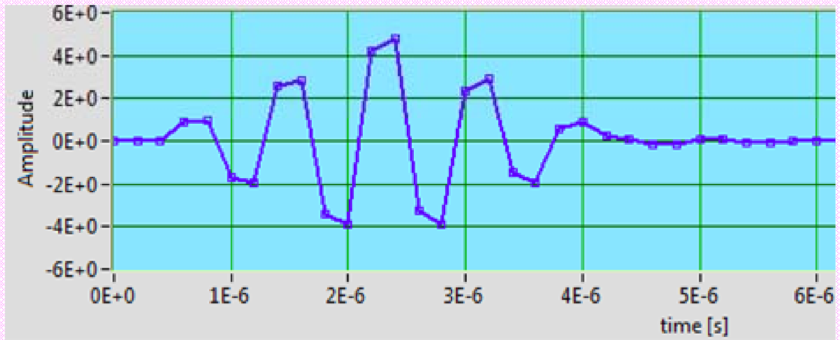
Génération d'une impulsion (1 seul échantillon de 1V)



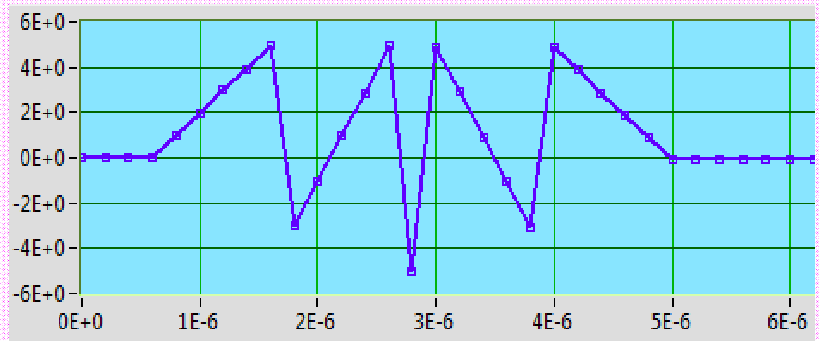
Filtres numériques par la transformation en Z : FIR (2)



$$Hfir(z) = a0 + a1 \cdot z^{-1} + a2 \cdot z^{-2} + \ldots\ldots + a20 \cdot z^{-20}$$



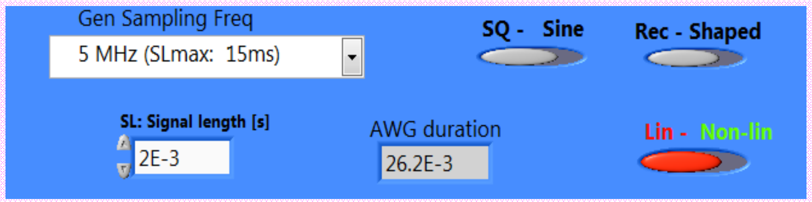
Déterminer a0 → a20 afin d'obtenir la réponse impulsionnelle suivante :



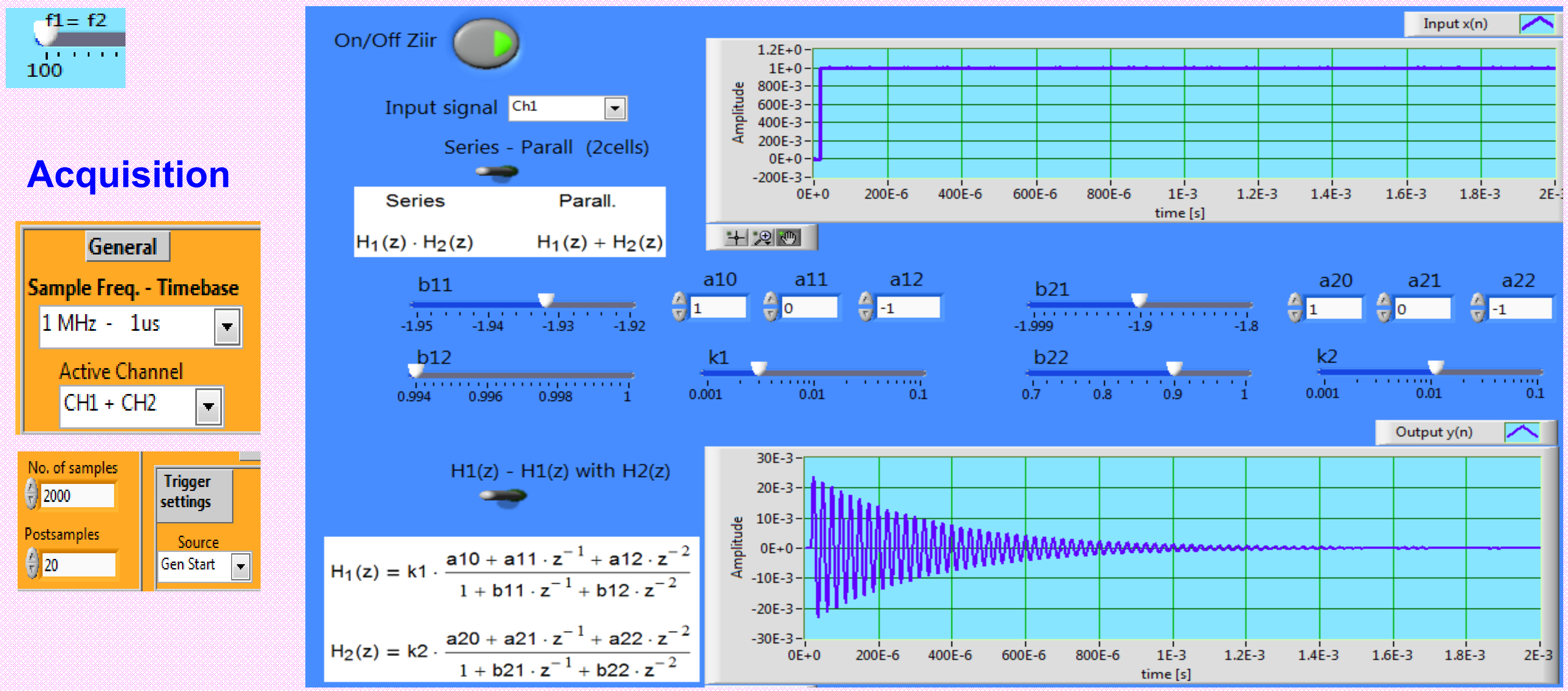
Filtres numériques par la transformation en Z : IIR (1)

Filtre passe-bande du 2ème ordre avec fs = 1 MHz, fr = 40 KHz et Bw = 1 KHz

Réponse à un saut unitaire



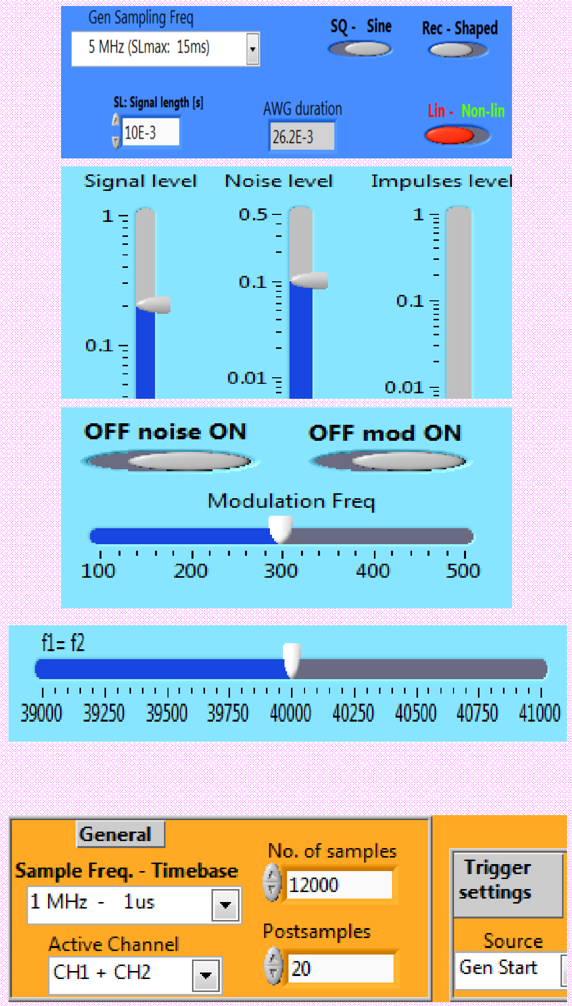
$$H(z) = 0.003 \cdot \frac{1 - z^{-2}}{1 - 1.932 \cdot z^{-1} + 0.994 \cdot z^{-2}}$$

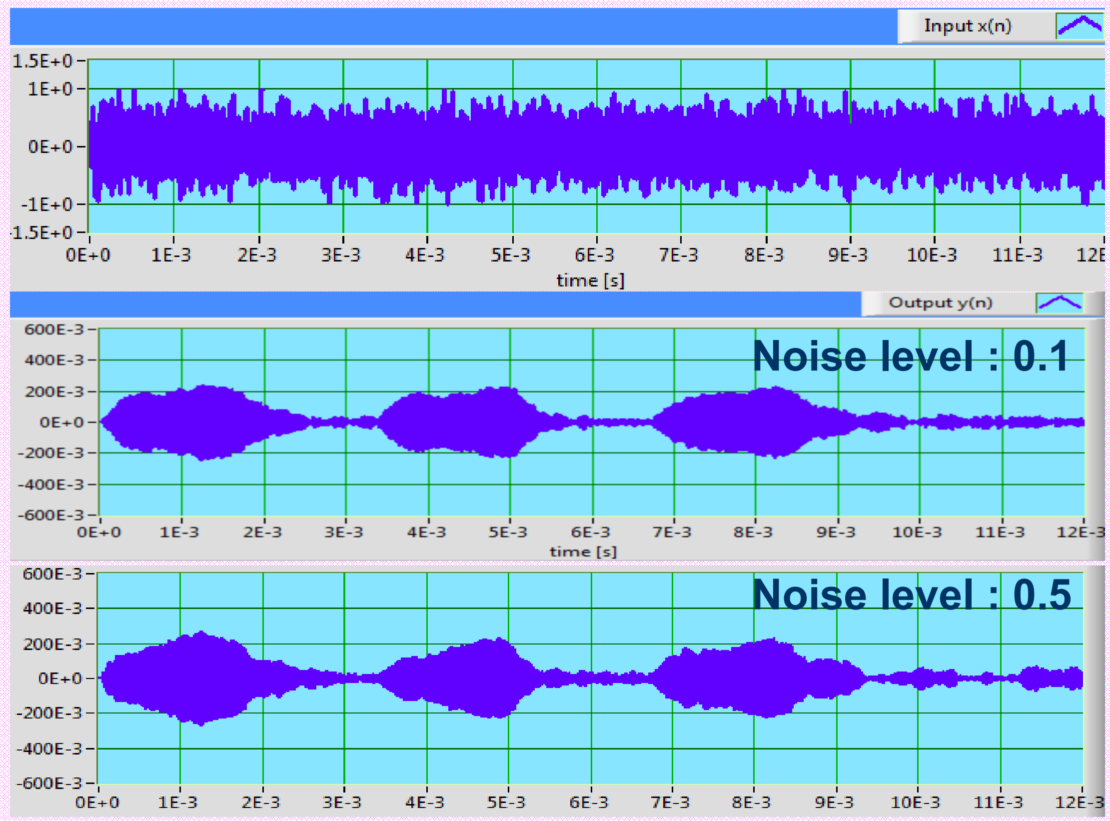


Analog and Digital Signal Processing

Filtres numériques par la transformation en Z : IIR (2)

Réduction de bruit gaussien additionné à un signal de 40 KHz modulé en tout-ou-rien (OOK)

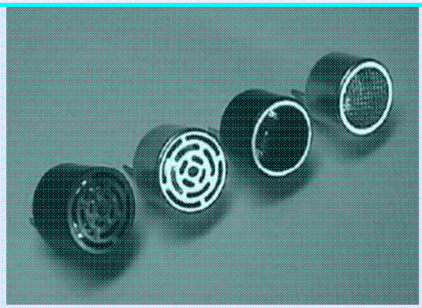




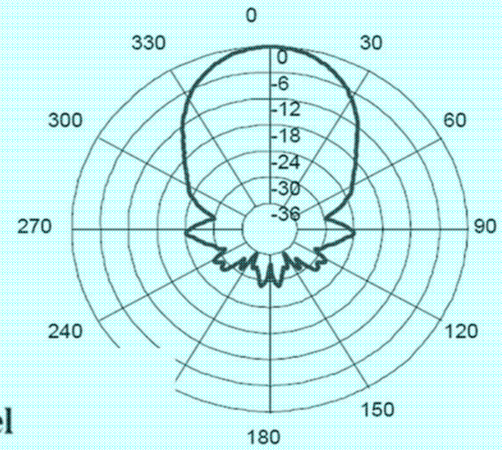
1. *Mettre **b_{12} = 0.998** et réajuster **b_{11}** afin que le filtre résonne toujours à la bonne fréquence. Que remarque-t-on ?*
2. *Tester la mise en série de deux filtres identiques.*

Ultrasons: Acquisition et traitement de signaux réels

Transducteurs à ultrasons de 40kHz



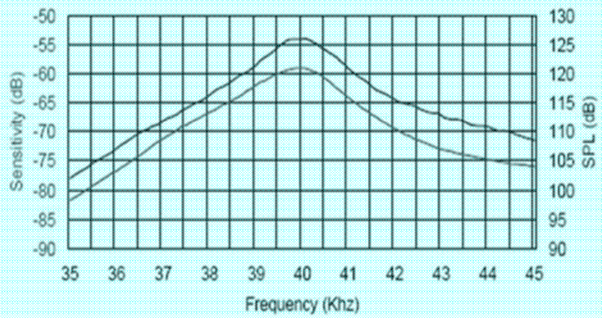
400ST160		Transmitter
400SR160		Receiver
Center Frequency		40.0±1.0Khz
Bandwidth (-6dB)	400ST160	2.0Khz
	400SR160	2.5Khz



Transmitting Sound Pressure Level at 40.0Khz; 0dB re 0.0002µbar per 10Vrms at 30cm	120dB min.
Receiving Sensitivity at 40.0Khz 0dB = 1 volt/µbar	-65dB min.
Capacitance at 1Khz ±20%	2400 pF
Max. Driving Voltage (cont.)	20Vrms
Total Beam Angle -6dB	55° typical
Operation Temperature	-30 to 80°C
Storage Temperature	-40 to 85°C

Sensitivity/Sound Pressure Level

Tested under 10Vrms @30cm



Beam Angle: Tested at 40.0Khz frequency

Ultrasons : Caract. électriques – Intro. tests pratiques

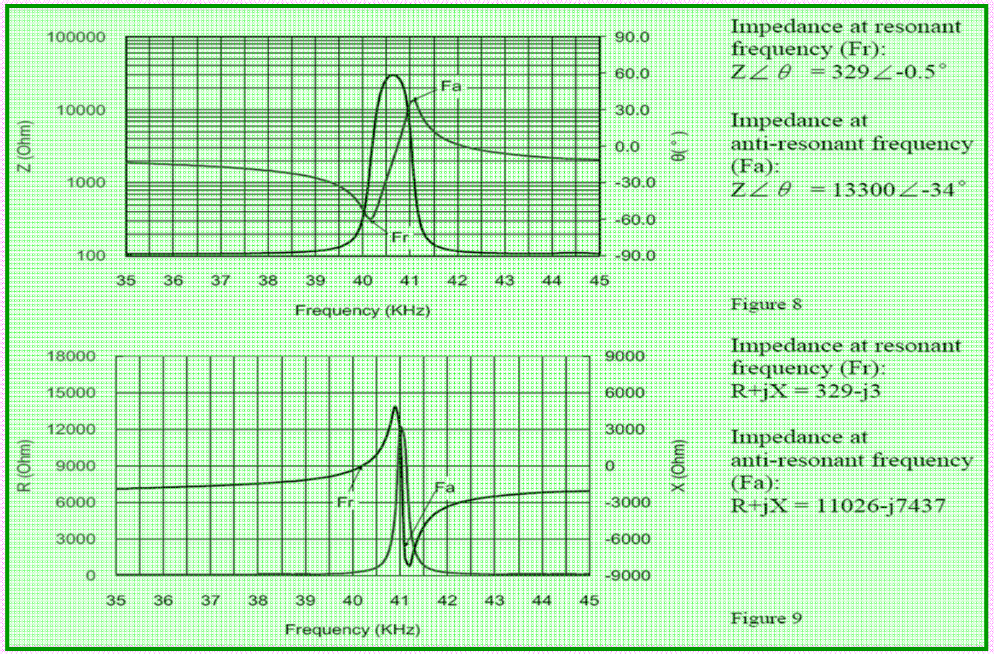
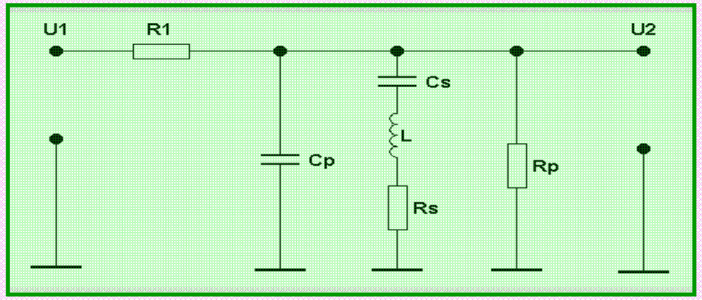


Schéma équivalant simplifié



Si ***Rs* = 0 Ω** et ***Rp* → ∞ →**
Z_{equi} = (Z_{cp} // (Z_{Cs}+Z_L). Alors :

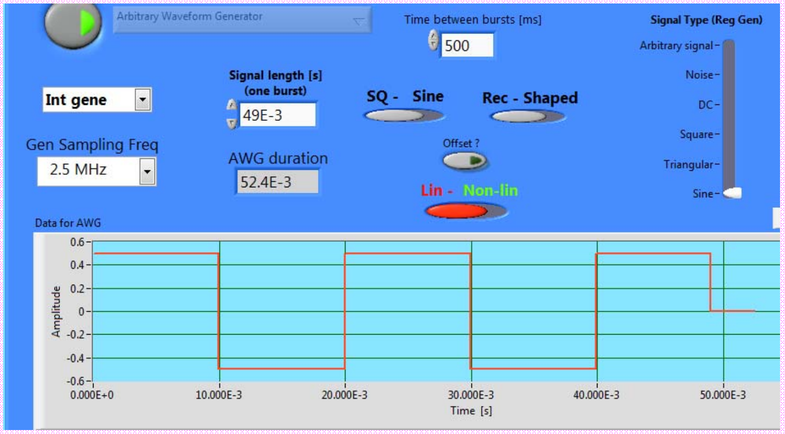
$$Z_{equi} = \frac{1}{j\cdot\omega\cdot Cp}\cdot\frac{1-\frac{1}{\omega^2\cdot L\cdot Cs}}{1-\frac{1}{\omega^2\cdot L\cdot Cx}}$$

Tests pratiques – Transducteurs de 40kHz 400ST and 400SR

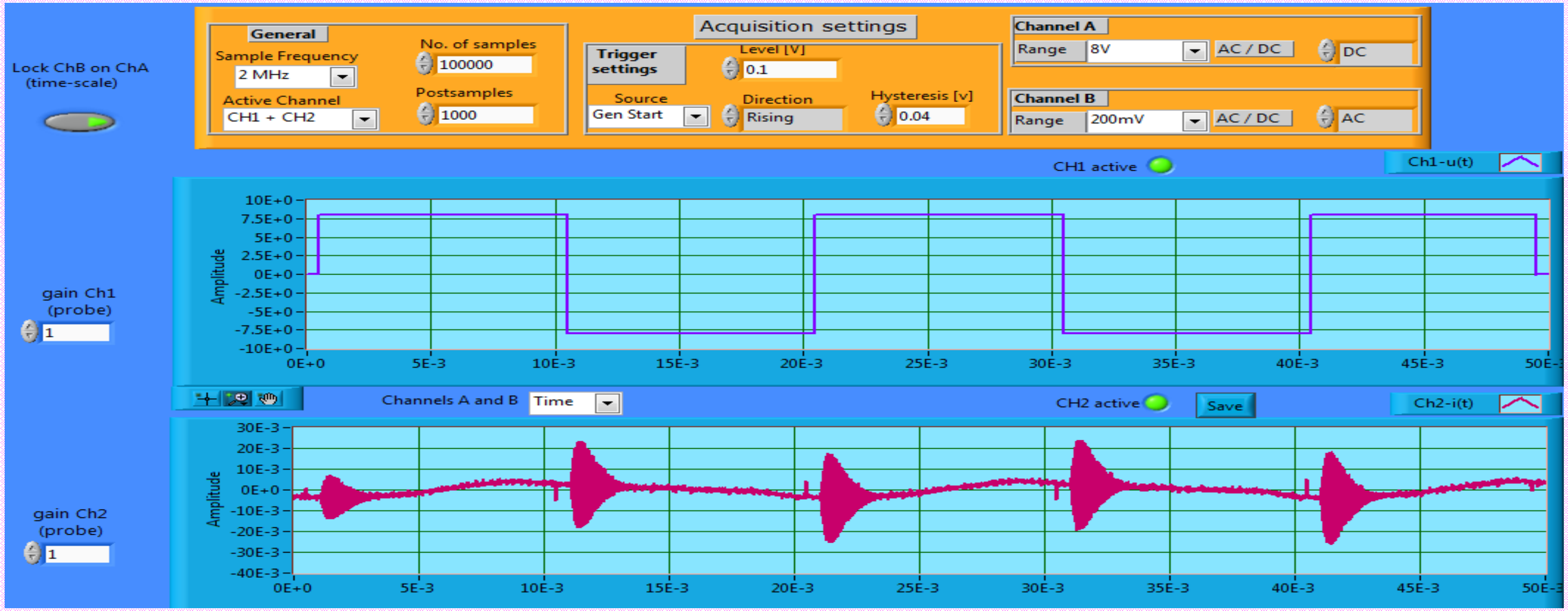
1. Mettre les deux transducteurs face-à-face à environ 20cm.
2. Trouver la fréquence qui produit le plus fort signal sur le transducteur de réception (35 kHz → 45 kHz).
1. Vérifier la directivité des transducteurs.
2. Déterminer la vitesse de propagation du son dans l'air.

Principle: $\lambda_{son} = v_{son}$ ***/ fréquence***

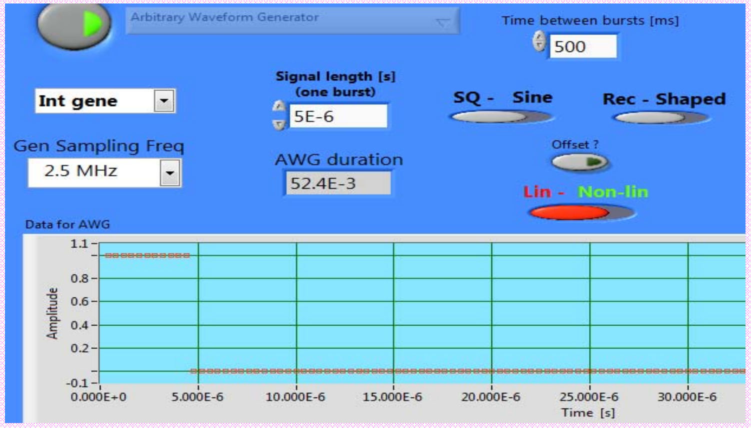
Ultrasons : Réponse à un saut unitaire



50Hz – 10V fmin = fmax

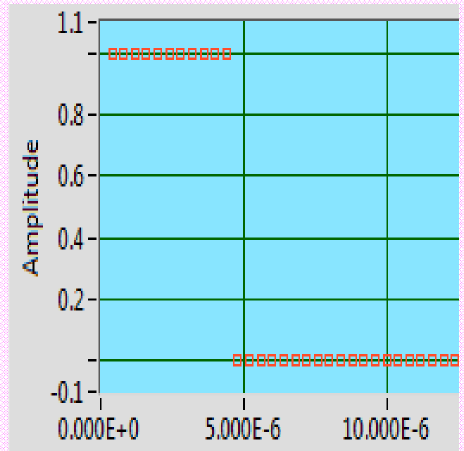


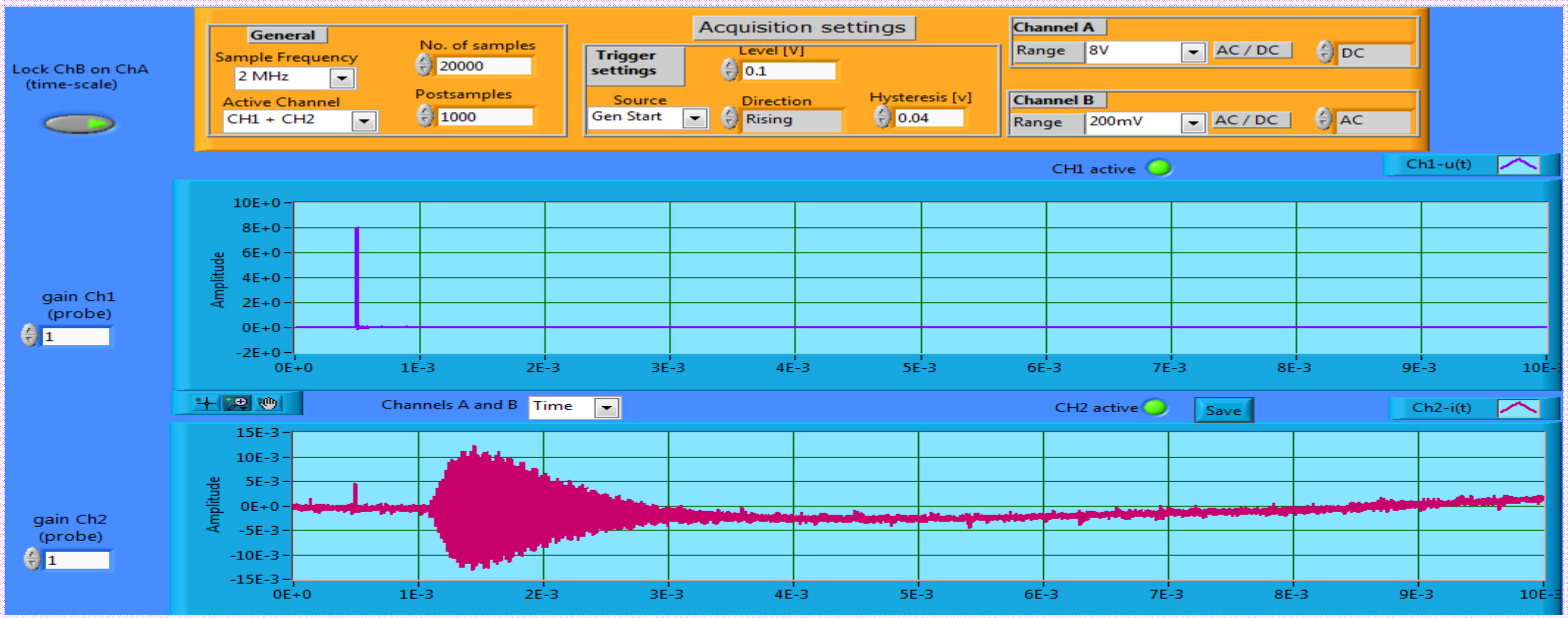
Ultrasons : Réponse impulsionnelle



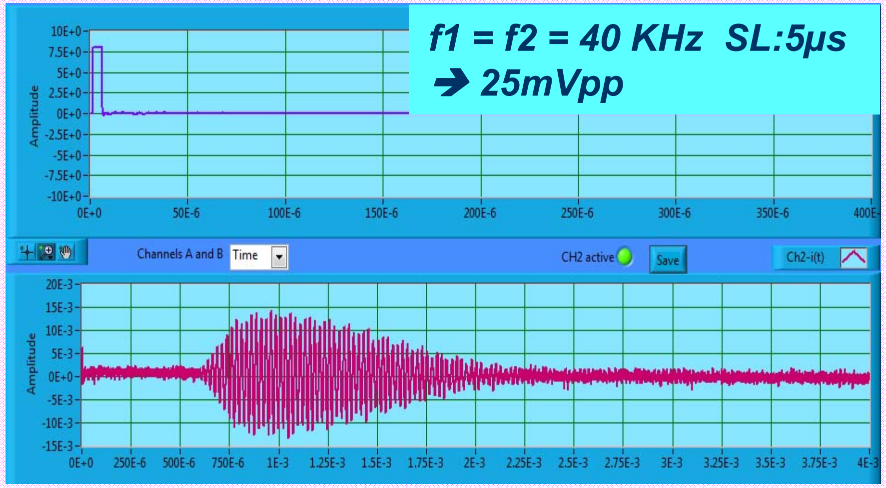
50Hz, 10V
fmin = fmax

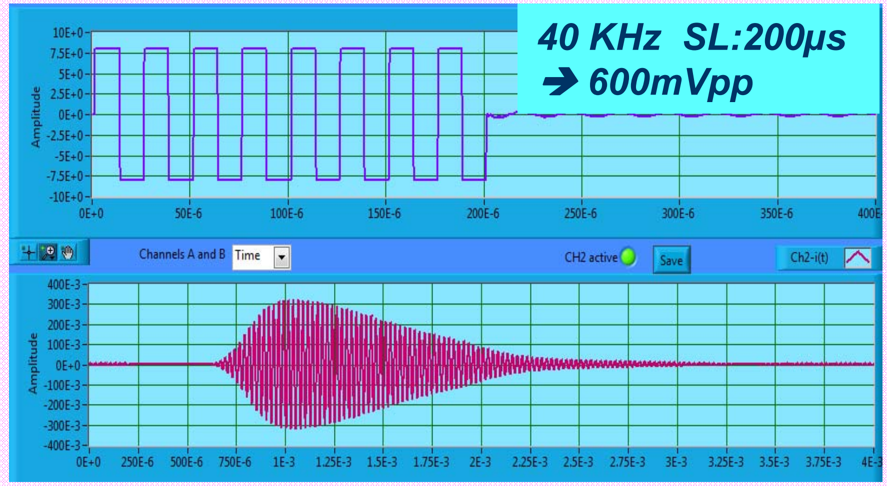
Pulse duration : 5µs

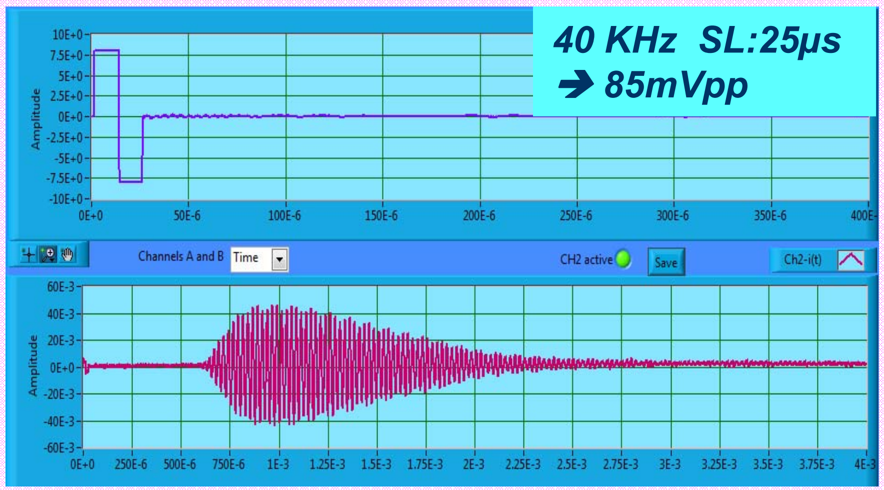


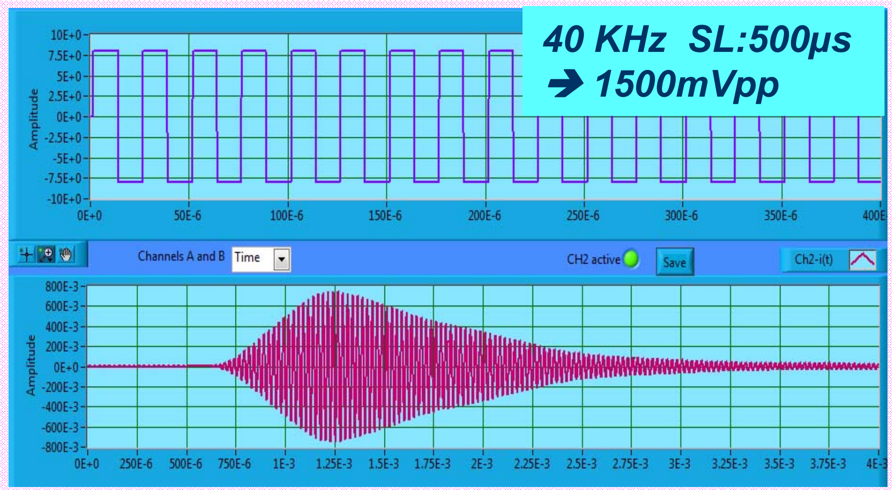


Ultrasons : Augmentation de l'amplitude de la réponse





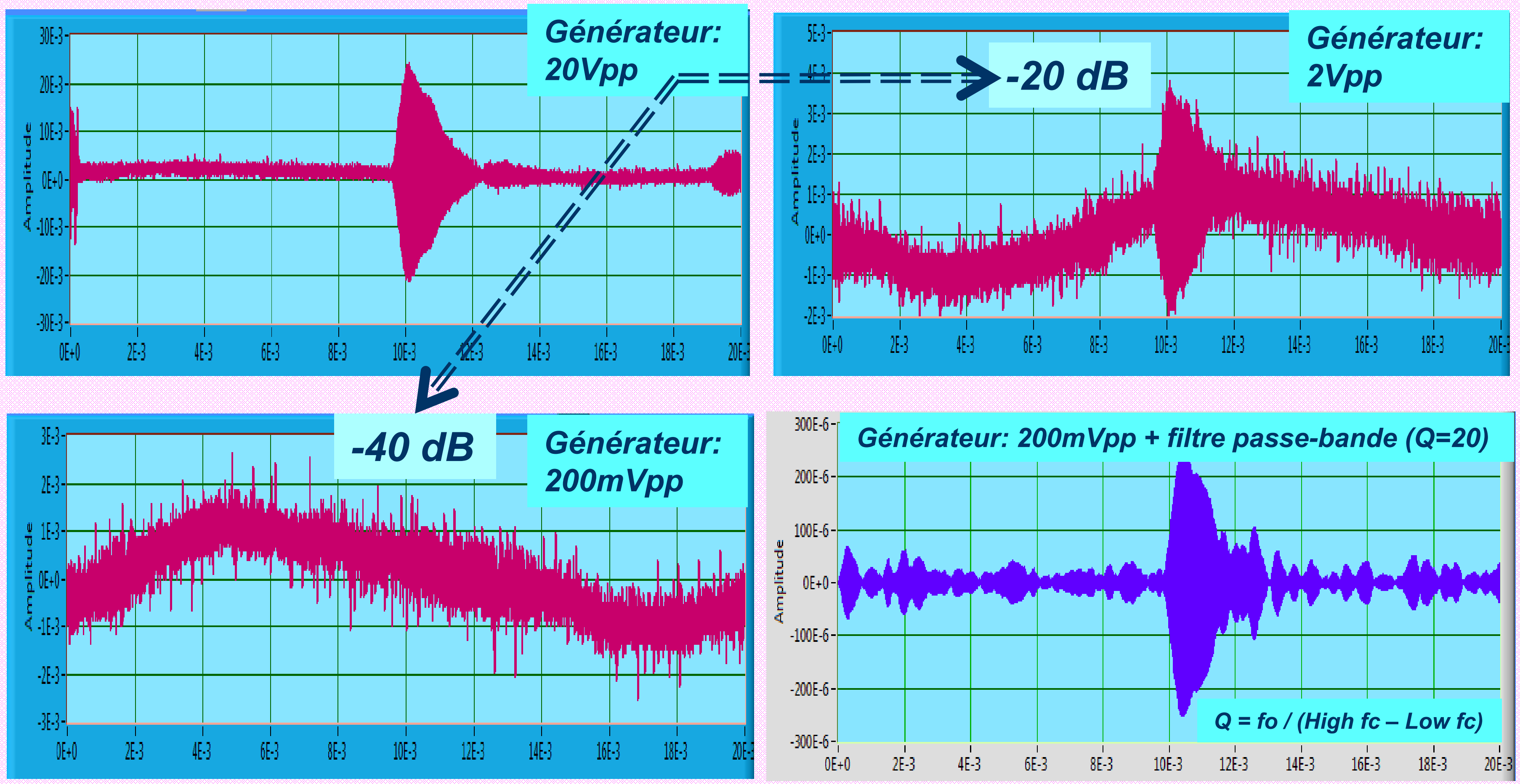




Ultrasons : Amélioration du rapport signal-sur-bruit (SNR)

Mesure de la hauteur du plafond: les 2 tranducteurs sont placés côte-à-côte face au plafond

Générateur: ~ 40kHz carré – durée du burst: 200 µs



Ultrasons : Mesures d'atténuations

Position initiale: 15cm entre les transducteurs *(page A4: 30cm x 21cm)*

Transducteurs: 40 kHz (piezo) - TiePie HS3 - LabView: TiePie HE-ARC 2014-10-22

Generator: Gen Sampling Freq: 5MHz, fmin=fmax~40kHz (choisi pour le maximum de signal reçu)

SQ , Rec, Signal length (one burst): 200μs, Amplitude : 10V, DC offset: 0V

Acquisition setting : Sample Frequency: 5MHz, 20000 samples, Trigger settings/Source: Gen Start

Filtering: Top filter, Bandpass: Low fc – High fc: à déterminer

Ultrasons : Mesures d'atténuations (suite)

Procédure: Vérifier la forme du signal reçu sans la feuille de papier entre les transducteurs, tourner légèrement une des planche en bois afin de réduire les réflexions. Mettre "Low fc" du filtre passe-bande à une fréquence d'environ 10% en dessous de celle des transducteurs et "High fc" à environ 10% en dessus, soit 36kHz et 44 kHz. Enregistrer la référence avec **Ref XCh2**, vérifier sa forme et sa position (**Display Gain = 1**). Insérer la feuille de papier entre les deux transducteurs (voir page précédente) et déterminer l'atténuation en ajustant le "**Display Gain**" afin que les amplitudes de la montée des signaux soient identiques. Ne pas oublier de réadapter "**ChB range**" au nouveau signal reçu qui est bien plus faible que précédemment. Afin de réduire le bruit, mettre **K = 5** (moyennage des acquisitions).