

# Typical stages in digital signal processing

## D/A Conversion

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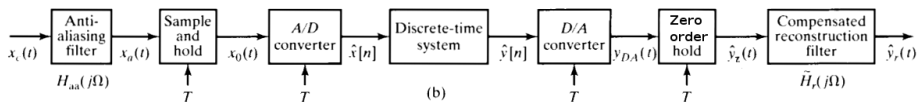
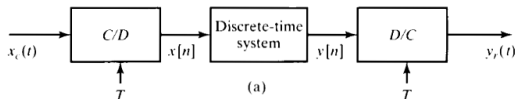
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- 1 D/A Conversion
- 2 Ideal D/A Converter
- 3 Zero-order holder
- 4 Reconstruction Filter
- 5 Reconstruction Filtering Strategies

- The DAC (Digital-Analog Converter) reverses the ADC process.
- It converts an abstract finite-precision number (usually a fixed-point binary number) into a physical quantity (voltage).

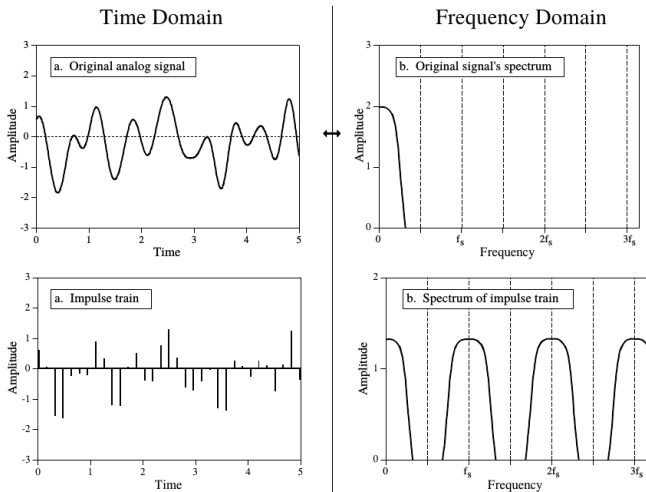
Some important DAC features:

- **Resolution:** the number of possible output levels the DAC is designed to reproduce. This is usually stated as the number of bits of the DAC.
- **Maximum sampling rate:** the maximum speed at which the DACs circuitry can operate and still produce correct output. *Su frecuencia define la frecuencia del tren de pulsos a su salida. A veces conviene una f mayor al conversor AD*



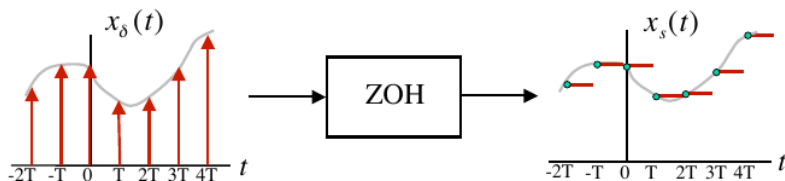
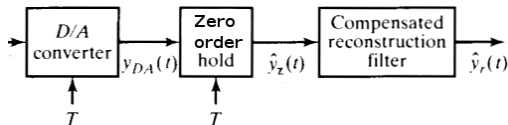
# Ideal D/A Converter

- DAC decodes the signal making a conversion from a bit sequence to an impulse train.
- The impulse train contains a duplication of analog signal spectrum.
- The original analog signal is reconstructed by passing this impulse train through a low-pass filter, with cutoff frequency equal to  $f_s/2$ .



# Zero-order holder

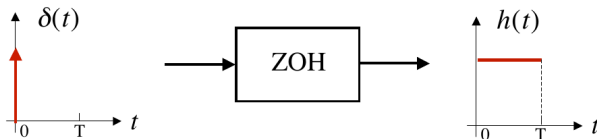
- Impulse train is a mathematical method pure.
- DACs operate by **holding** the last value until another sample is received.
- **Zero-order holder** interpolates analog values between times  $T$ ,  $2T$ ,  $3T$ , ..., and produces the staircase appearance.



Sostiene el impulso para hacerlo un escalón de tiempo  $T$

## Transforma impulsos en pulso

The impulse response of a zero-order holder.



In the frequency domain:

T de Fourier

$$H_{\text{ZOH}}(f) = \mathcal{F} \left\{ \text{rect} \left( \frac{t - T_s/2}{T_s} \right) \right\} = \frac{\sin(\pi f / f_s)}{(\pi f / f_s)} = \text{sinc}(\pi f / f_s) \quad (1)$$

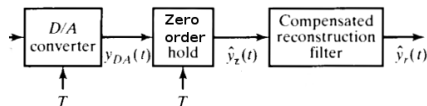
funcion rectangular

seno cardinal

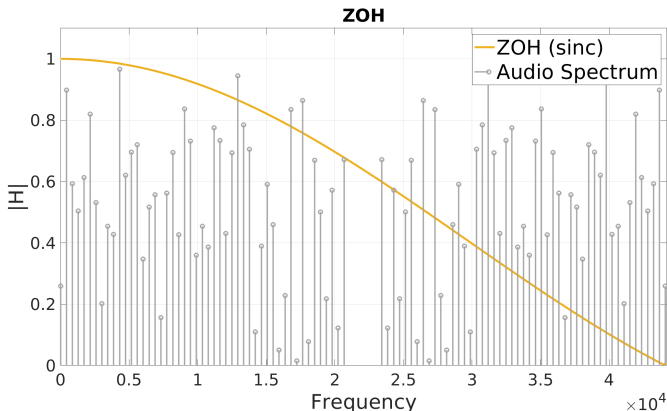
**sinc**: sine cardinal function.

# Zero-order holder, frequency analysis

- ZOH is the convolution of the impulse train with a rectangular pulse.
- In the frequency domain, ZOH Fourier transform (sinc) is being multiplied by the impulse train spectrum.



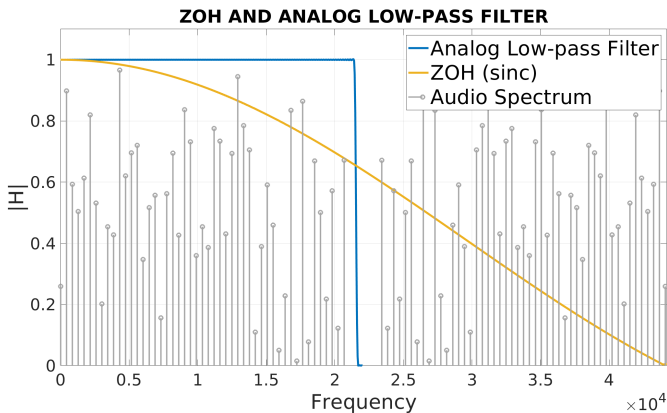
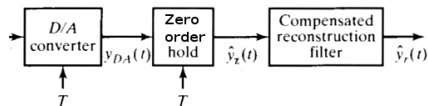
Example for audio:  $f_N = 22050$  Hz,  $f_s = 44100$  Hz.



Esta respuesta es un mal filtro pasa bajo.

## Zero-order holder, frequency analysis, 2

An analog low-pass filter is required to cut frequencies  $> 22010$  Hz.



Convolution de esos elementos es producto en dom de las  $f$

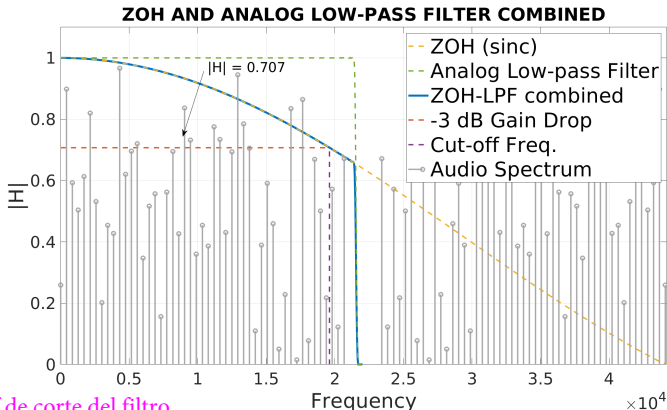
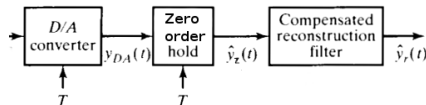
Hay que quitar el reflejo



# Zero-order holder, frequency analysis, 3

- The ZOH spectrum (sinc) produces a gain drop of -3 dB.
- The cut-off frequency is decreased to  $\approx 20$  kHz.

-3dB gain drop  
es la perdida de  
la mitad de la  
potencia. En ese  
punto se define  $f$   
de corte



Linea en  
azul es  
respuesta  
efectiva.  
Producto  
de las  
anteriores

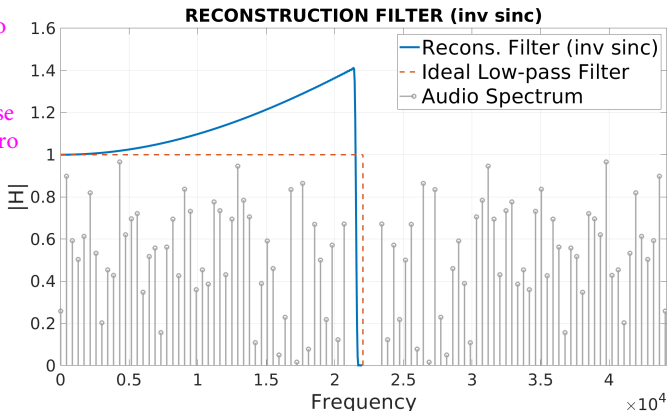
Modifica la  $f$  de corte del filtro

# Reconstruction Filter

- RF removes all frequencies above  $f_s/2$ .
- It boosts the frequencies by the reciprocal of the zeroth-order holder's effect.
- It is also known as invert sinc filter (inv sinc).

$$H_{RF}(f) = \frac{(\pi f / f_s)}{\sin(\pi f / f_s)} = 1 / \text{sinc}(\pi f / f_s) \quad (2)$$

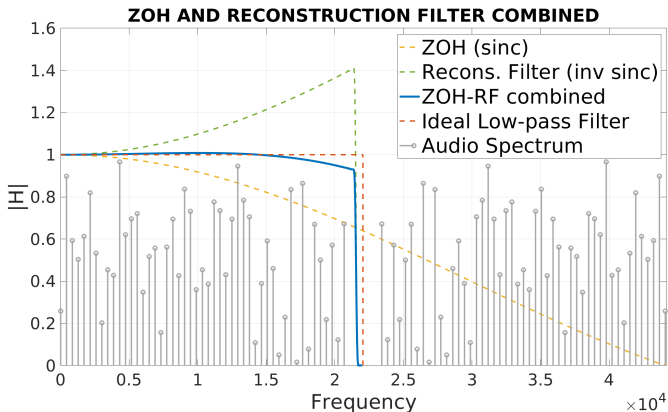
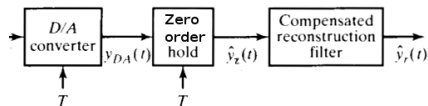
Se saca filtro  
analógico  
común y se  
pone un filtro  
especial



Es la inversa  
del seno  
cardinal. Da  
mas  
ganancia al  
aumentar la  
frecuencia

## Reconstruction Filter, 2

The reconstruction filter compensates the gain drop.



# Reconstruction Filtering Strategies

- 1) **Ignore** the effect of the zero-order holder and **accept** the consequences.
- 2) **Pre-equalizing**: digital filter to remove the sinc effect [3].
- 3) **Post-equalizing**: analog filter to remove the sinc effect [3].

Formas de  
implementar el  
filtro de  
reconstrucción

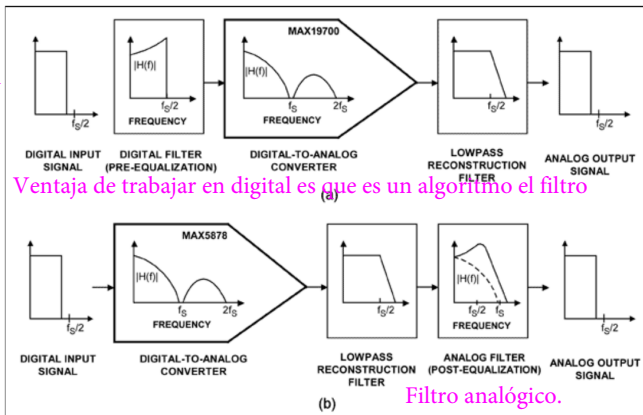
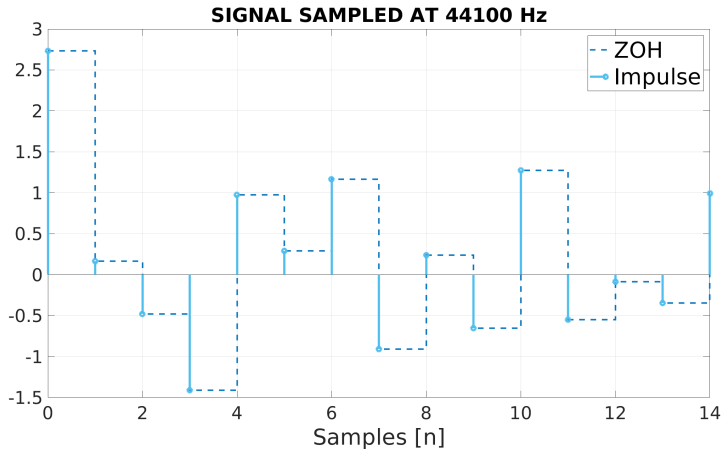


Figure 4. A pre-equalization digital filter is used to cancel the effect of sinc rolloff in a DAC (a). As an alternative, you can use a post-equalization analog filter for the same purpose (b).

# Reconstruction Filtering Strategies, Oversampling

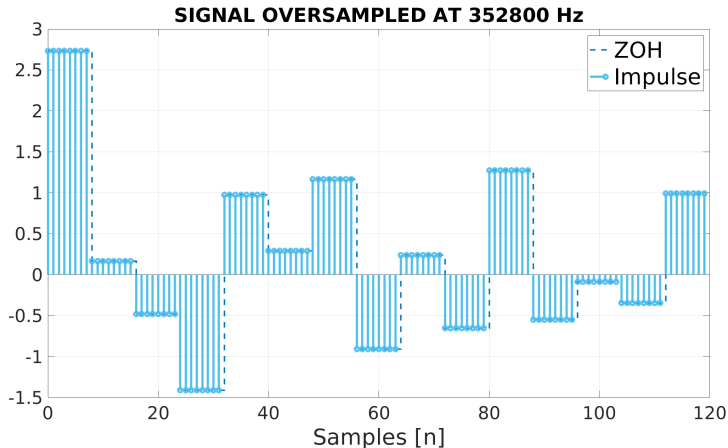
## 4ta opcion de implementacion del filtro

In CD players, data sampling rate is 44.1 kHz. Se implementa en estos reproductores esta solucion



# Reconstruction Filtering Strategies, Oversampling, 2

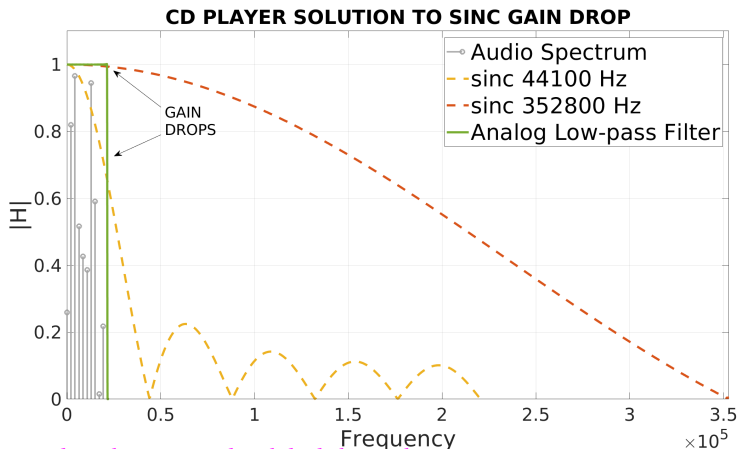
Data is oversampled by a factor of 8 to 352.8 kHz.



No da información, pero da pulso más pequeño a la salida del zoh. Esto en el dom de la frecuencia se va a transformar en un seno cardinal más ancho.

# Reconstruction Filtering Strategies, Oversampling, 3

- At 80 % of Nyquist frequency, the output amplitude is attenuated by 2.42dB.
- Distortion for ZOH is effectively eliminated by using oversampling.



Casi no hay caída en la potencia de salida de la señal

- 1 Alan V. Oppenheim and Ronald W. Schaffer. *Discrete-time signal processing*, 3rd Ed. Prentice Hall. 2010. Section 4.3.
- 2 Steven W. Smith. The Scientist and Engineer's Guide to Digital Signal Processing. Chapter 3, ADC and DAC. [Link](#).
- 3 Maxim Integrated. Equalizing Techniques Flatten DAC Frequency Response. Application Note 3853. August 2012.