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# **Basic Multirate Building Blocks**

A. Petraglia Universidade Federal do Rio de Janeiro DEL, COPPE

# **Introductory Remarks**

#### **Multirate Signal Processing**

- Substantial knowledge has been developed in digital multirate signal processing, specially in the early 1980s, when digital signal processing algorithms were becoming increasingly powerful;
- By that time, advances in IC technologies were making it possible the realization of DSPs capable of performing billions of operations per second;
- The motivation was to adjust the signal sampling rate according to its bandwidths variations from the input to the output of the network;
- Efficient structures have been derived to exploit this idea, as for example, the celebrated subband coding scheme, by which the signal is split into a number of frequency bands, and appropriately coded inside each band.

# **Introductory Remarks**

#### **Analog Multirate Signal Processing**

- The multirate approach is particularly important for analog networks to relax amplifier settling time requirements and reduce capacitance spread, thereby reducing power consumption and chip area;
- High quality CMOS capacitances and switches have made switchedcapacitor (SC) networks capable of incorporating the multirate techniques originally developed in digital signal processing;
- With the multirate option another degree of freedom can be obtained in the design of analog networks.

#### Analog Down-Sampler





*M* is a positive integer greater than 1.

 $\omega$ 

Sample-And-Hold Effect



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$$U(j\omega) = X(j\omega)S(j\omega)$$

$$X(j\omega) = \frac{\omega_s}{2\pi} \sum_{k=-\infty}^{\infty} A(j\omega - jk\omega_s)$$

$$S(j\omega) = \frac{2\pi}{\omega_s} \operatorname{sinc}\left(\frac{\omega}{\omega_s}\right) e^{-j\pi\omega/\omega_s}$$



#### Sample-And-Hold Effect



$$V(j\omega) = \left(\frac{\omega_s}{2\pi M} \sum_{k=-\infty}^{\infty} A\left(j\omega - j\frac{k\omega_s}{M}\right)\right) S'(j\omega)$$

$$S'(j\omega) = \frac{2\pi M}{\omega_s} \operatorname{sinc}\left(\frac{\omega M}{\omega_s}\right) e^{-j\pi\omega M/\omega_s}$$

- The down sampler must be preceded by a sampled-data lowpass filter with cutoff frequency  $\omega_s/2M$  to prevent aliasing.

#### Analog Up-Sampler



 Narrows every sample by factor L, leaving L – 1 zero-valued samples between every 2 adjacent samples.

Sample-And-Hold Effect



$$Y(j\omega) = \left(\frac{\omega_s}{2\pi} \sum_{k=-\infty}^{\infty} A(j\omega - jk\omega_s)\right) S''(j\omega)$$
$$S''(\omega) = \frac{2\pi}{L\omega_s} \operatorname{sinc}\left(\frac{\omega}{L\omega_s}\right) e^{-j\pi\omega/L\omega_s}$$









#### Sample-And-Hold Effect





- The L 1 images of the input spectrum carry redundant information;
- Can be removed by a sampled-data lowpass filter with cutoff frequency  $\omega_s/2$ .

Demultiplexer



(a) Functional diagram; (b) Equivalent block diagram.

**Multiplexer** 



(a) Functional diagram; (b) Equivalent block diagram.

#### The Nobel Identities

- Often lead to efficient realizations of multirate systems;
- Enable us to "move around" certain building blocks;
- Useful in design and analysis of multirate systems.

#### (I) For Down-samplers

$$\bullet \bullet \downarrow M \longrightarrow H(z) \longrightarrow = \bullet \bullet H(z^M) \longrightarrow \downarrow M \longrightarrow$$



#### (II) For Up-samplers





### **Decimation and Interpolation Filters**

#### **Decimation Filter**

$$V_{in} \underbrace{\downarrow}_{\omega_s} H(z) \underbrace{\downarrow}_{\omega_s} \underbrace{\downarrow}_{\omega_s} M \underbrace{\downarrow}_{\omega_s/M} V_{out}$$

Ideal Frequency Response

$$H(e^{j\omega}) = \begin{cases} 1, & |\omega| \le \omega_s/2M \\ 0, & \omega_s/2M < |\omega| \le \omega_s/2 \end{cases}$$

- The cascade of a lowpass filter and a down-sampler is called a *decimation filter*;
- Filter H(z) attenuates input frequency components greater than  $\omega_s/2M$ ;
- The sampling rate can then be reduced by the factor *M*.

### **Decimation and Interpolation Filters**

#### **Interpolation Filter**

$$V_{in} \xrightarrow{\uparrow} L \xrightarrow{\uparrow} G(z) \xrightarrow{\searrow} V_{out}$$
$$\omega_s \qquad L\omega_s \qquad L\omega_s$$

Ideal Frequency Response

$$G(e^{j\omega}) = \begin{cases} 1, & |\omega| \le \omega_s/2\\ 0, & \omega_s/2 < |\omega| \le L\omega_s/2 \end{cases}$$

- The cascade association of an upsampler with a lowpass filter is called an *interpolation filter*;
- The zero-valued samples are filled in by the lowpass filter, hence implementing interpolation;
- In the frequency domain L 1 copies of the input spectrum are created between consecutive multiples of the input sampling frequency.

### **Sampling of Analog Signals**





## Sampling of Analog Signals



- Output of the anti-aliasing filter (AAF) is oversampled by factor M:
  → transition band of AAF can be increased;
- Accurate bandlimiting provided by SC filter H(z) allows sampling rate reduction:
  - $\rightarrow$  reduction of ADC and DSP power consumption.

### **Reconstruction of Analog Signals**



- Interpolation filters are useful in alleviating the specifications of reconstruction filters, H<sub>r</sub>(s), that follow D/A converters;
- The SC filter H(z) removes the L 1 images produced by the up-sampler;
- $H_r(s)$  has thus the easier task of selecting (smoothing) the baseband components of  $V_{in}$ .