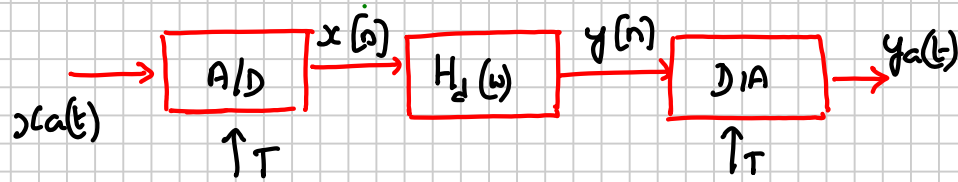


## ECE 310 : Lecture 27 : Multi-Rate Signal processing

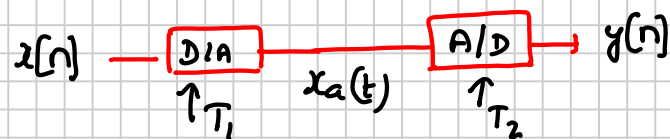
DSP system:



Note that the sampling rate across the system is same. Some applications require that a signal of sampled at rate  $T_1$  be converted to an equivalent signal sampled at rate  $T_2$ .

### SAMPLING RATE CONVERSION:

① Convert signal to continuous time and resample.



①  $T_1$  and  $T_2$  can be arbitrarily chosen

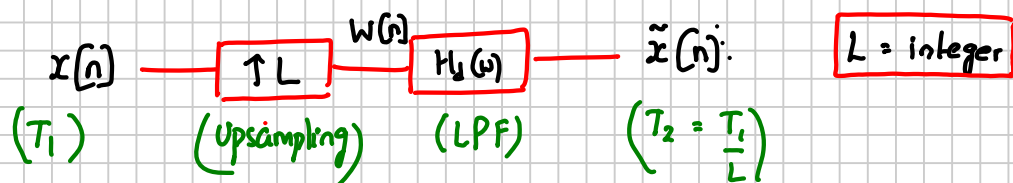
② Distortion in D/A and A/D Converters

We will look at sampling rate conversion by an integer factor in the digital domain.

### ① Increasing Sampling Rate (Interpolation):

$$T_2 = \frac{T_1}{L}$$

Sampling Rate can be increased by UPSAMPLING followed by LOW PASS FILTERING



Upsampling in Digital domain:

$$x[n] \xrightarrow{\uparrow L} w[n]$$

$$w[n] = \begin{cases} x[n/L] & n = \text{multiple of } L \\ 0 & \text{otherwise} \end{cases}$$

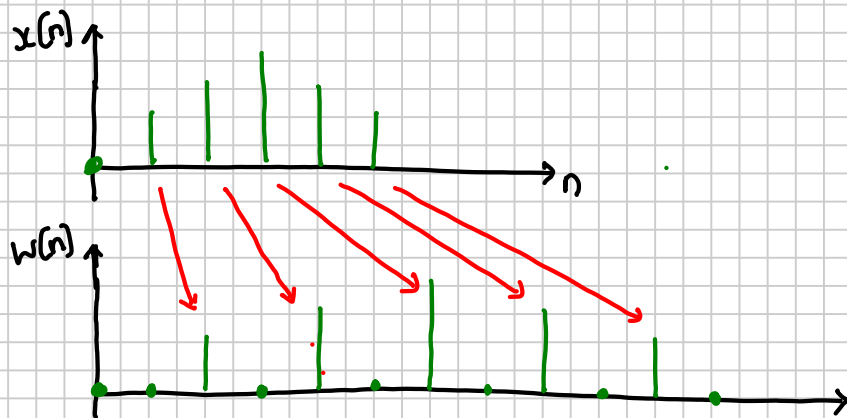
Note: Upsampler inserts  $(L-1)$  zeros between two samples of  $x[n]$

Example:  $L=2$  upsampler

$$w[n] = \begin{cases} x[n/2] & n = \text{multiple of } 2 \\ 0 & \text{otherwise} \end{cases}$$

$$w[0] = x[0] \quad w[1] = 0 \quad w[2] = x[1]$$

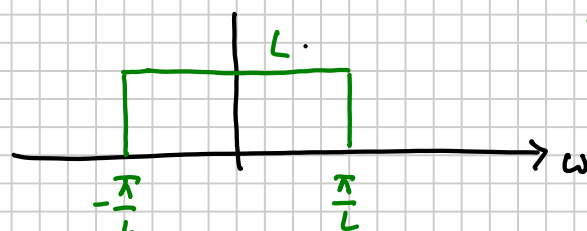
$$w[3] = 0 \quad w[4] = x[2]$$



Step 2: Convert the zero-valued samples into interpolated samples. This is achieved by a LPF at the output of upsampler.

$$x[n] \xrightarrow{\uparrow L} w[n] \xrightarrow{H_d(\omega)} \tilde{x}[n]$$

$H_d(\omega)$  is an ideal LPF:



Note:  $L = \frac{T_1}{T_2}$

To see the above let us look at the problem in Fourier domain.

$$\sum_{n=-\infty}^{\infty} w[n] e^{-j\omega n} = \sum_{\substack{n = \text{mult. of } L \\ (n = kL)}} w[n] e^{-j\omega n}$$

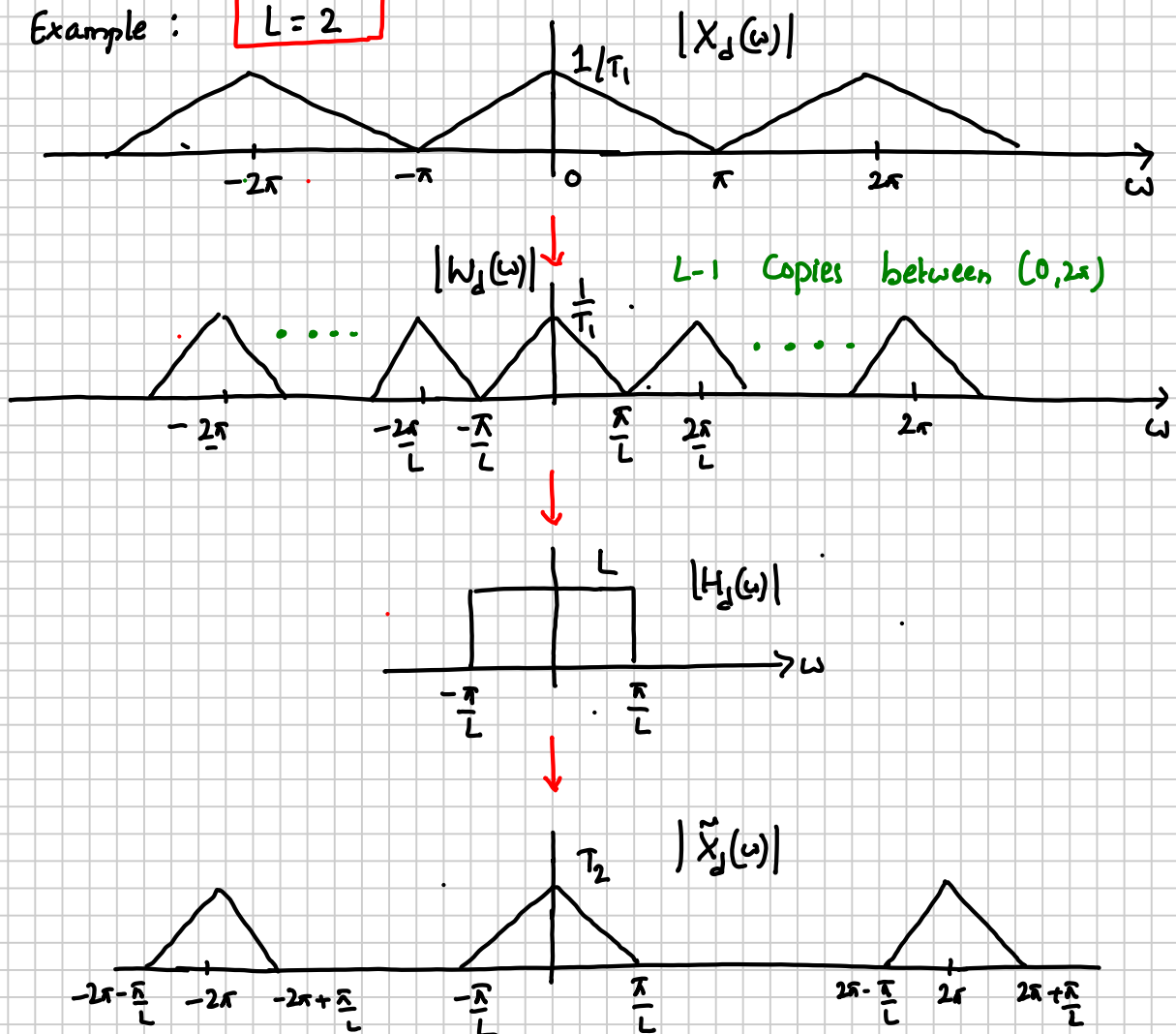
$$= \sum_{k=-\infty}^{\infty} w[kL] e^{-j\omega kL}$$

$$= \sum_{k=-\infty}^{\infty} x[k] e^{-j\omega L k}$$

$$\Rightarrow \boxed{W_d(\omega) = X_d(\omega L)}$$

$\Rightarrow$  Spectrum of  $w[n]$  is a  $L$ -fold Compressed version of  $x[n]$

Example:  $\boxed{L=2}$



②

$L=3$

