



ET 644 – Advanced Digital Signal Processing

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Department of Engineering and Technology
zhang@email.wcu.edu

Session 9 DAC

Outline:

- Sample and Hold (S & H), Reconstruction - section 9.3.1
- Digital Up-Sampling (Section 10.3)
- Efficient Up-sampling (Section 10.5.2)
- Reminder: Exam 1 covers up to this class.

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- See Fig. 9.22 on p.767 \implies S/H reconstruction
- Digital Up-Sampling (Increasing sampling rate just prior to D/A conversion)

$$x[n] = x_a\left(\frac{n}{F_s}\right) \longrightarrow \boxed{\text{DT System}} \longrightarrow y[n] = x_a\left(\frac{n}{LF_s}\right)$$

- where $L = \text{integer}$
- i.e., increasing sampling rate by “L” just prior to S/H.
- e.g., A CD player advertised “8 times over sampling” meaning $L = 8$.

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- Mathematical Preamble:

$$x[n] * \delta[n - n_0] = x[n - n + 0]$$

- In particular, if $n_0 = 0$

$$x[n] = \sum_{k=-\infty}^{\infty} x[k] \delta[n - k]$$

- Consider: $\sum_{k=-\infty}^{\infty} x[k] \delta[n - 2k]$
 $= \cdots + x[-1] \delta[n + 2] + x[0] \delta[n] + x[1] \delta[n - 2] + \cdots$
 $= \{ \cdots, 0, x[-1], 0, x[0], 0, x[1], 0, \cdots \}$
- See graphical representation of the system.



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- More generally:
- Insert L-1 zeros between each pair of samples



- End of Preamble.
- As long as $F_s > 2W$, (i.e., $|X_a(F)| \approx 0$, for $|F| > W$)
- Then $x_a(t)$ maybe reconstructed as:

$$x_a(t) = \sum_{k=-\infty}^{\infty} x[k] h_{LP}(t - kT_s)$$

- See graphical presentation.

$$\begin{aligned} y[n] &= x_a\left(n\frac{T_s}{L}\right) = x_a\left(\frac{n}{LF_s}\right) \\ &= \sum_{k=-\infty}^{\infty} x[k] h_{LP}\left(\underbrace{n\frac{T_s}{L} - kT_s}_{(n-kL)\frac{T_s}{L}}\right) \end{aligned}$$

$$= \sum x[k] h_{LP}[n - kL]$$

where $h_{LP}[n] = h_{LP}\left(n\frac{T_s}{L}\right) = h_{LP}\left(\frac{n}{LF_s}\right)$.

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$$h_{LP}[n - kL] = h_{LP}[n] * \delta[n - kL]$$

$$y[n] = \left\{ \sum_{k=-\infty}^{\infty} x[k] \delta[n - kL] \right\} * h_{LP}[n]$$

- Convolution is distributive

$$x[n] = x_a\left(\frac{n}{F_s}\right) \longrightarrow \left(\uparrow L \right) \longrightarrow y[n] = x_a\left(\frac{n}{LF_s}\right)$$

- Now let's investigate the specs of this low-pass filter.
- Example: for speech with bandwidth $\approx 4\text{kHz}$, sampled at $F_s = 12\text{kHz}$.
- Desire to increase sampling rate by $L=2$ to 24 kHz.
- $\omega_p = 2\pi \frac{4}{2(12)} = \frac{\pi}{3}$
- $\omega_s = 2\pi \frac{12-4}{2(12)} = \frac{2\pi}{3}$
- See demo `upsamplex2eg1.m`

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- Efficient Digital Upsampling

- $L = 2$ for illustration

$$y[n] = \sum_{k=-\infty}^{\infty} x[k] h_{LP}[n - 2k]$$

- Consider $y[2n]$ and $y[2n+1]$ separately.

$$y_0[n] = y[2n] = \sum_{k=-\infty}^{\infty} x[k] h_{LP}[\underbrace{2n - 2k}_{2(n-k)}] = x[n] * h_0[n]$$

- where: $h_0[n] = h_{LP}[2n]$

$$x[n] \longrightarrow \boxed{h_0[n] = h_{LP}[2n]} \longrightarrow y_0[n] = y[2n]$$

- Similarly,

$$y_1[n] = y[1 + 2n] = \sum_{k=-\infty}^{\infty} x[k] h_{LP}[\underbrace{1 + 2n - 2k}_{1+2(n-k)}] = x[n] * h_1[n]$$

- where: $h_1[n] = h_{LP}[1 + 2n]$

- See graphical presentation of the “commutator.” (see demo: upsamlex2eg2.m)