Arbitrary-Rate Sampling Rate Converter

- The estimation of a discrete-time signal value at an arbitrary time instant between a consecutive pair of known samples can be solved by using some type of interpolation
- In this approach an approximating continuous-time signal is formed from a set of known consecutive samples of the given discrete-time signal

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Arbitrary-Rate Sampling Rate Converter

- The value of the approximating continuoustime signal is then evaluated at the desired time instant
- This interpolation process can be directly implemented by designing a digital interpolation filter

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Ideal Sampling Rate Converter

- Let the impulse response of the analog lowpass filter is denoted by $g_a(t)$
- Then the output of the filter is given by $\hat{x}_a(t) = \sum_{\ell=-\infty}^{\infty} x[\ell] g_a(t - \ell T)$
- If the analog filter is chosen to bandlimit its output to the frequency range $F_g < F_T'/2$, its output $\hat{x}_a(t)$ can then be resampled at the rate F_T'

Ideal Sampling Rate Converter

- Since the impulse response $g_a(t)$ of an ideal lowpass analog filter is of infinite duration and the samples $g_a(nT'-\ell T)$ have to be computed at each sampling instant, implementation of the ideal bandlimited interpolation algorithm in exact form is not practical
- Thus, an approximation is employed in practice

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Ideal Sampling Rate Converter

- Problem statement: Given $N_2 + N_1 + 1$ input signal samples, x[k], $k = -N_1,...,N_2$, obtained by sampling an analog signal $x_a(t)$ at $t = t_k$ $= t_0 + kT_{in}$, determine the sample value $x_a(t_0 + kT_{in}) = y[\alpha]$ at time instant $t' = t_0 + kT_{in}$ where $-N_1 \le \alpha \le N_2$
- Figure on the next slide illustrates the interpolation process by an arbitrary factor

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Lagrange Interpolation Algorithm

- <u>Example</u> Design a fractional-rate interpolator with an interpolation factor of 3/2 using a 3rd-order polynomial approximation with $N_1 = 2$ and $N_2 = 1$
- The output y[n] of the interpolator is thus computed using $y[n] = P_{\alpha}(\alpha)x[n-2] + P_{\alpha}(\alpha)x[n-1]$

$$y_{[n]} - r_{-2}(\alpha)x_{[n-2]} + r_{-1}(\alpha)x_{[n-1]} + P_0(\alpha)x_{[n]} + P_1(\alpha)x_{[n+1]}$$

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11







- From the figure on the previous slide it can be seen that the value of α for computation of y[n], to be labeled α₀, is 0
- Substituting this value of α in the expressions for the Lagrange polynomial coefficients derived earlier we get

 $P_{-2}(\alpha_0) = 0, P_{-1}(\alpha_0) = 0$ $P_0(\alpha_0) = 1, P_1(\alpha_0) = 0$

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14

 Lagrange Interpolation Algorithm
The value of α for computation of y[n+1], to be labeled α₁, is 2/3
Substituting this value of α in the expressions for the Lagrange polynomial coefficients we get
P₋₂(α₁) = 0.0617, P₋₁(α₁) = -0.2963 P₀(α₁) = 0.7407, P₁(α₁) = 0.4938

Lagrange Interpolation Algorithm • The value of α for computation of y[n+2], to be labeled α_2 , is 4/3 • Substituting this value of α in the expressions for the Lagrange polynomial coefficients we get $P_{-2}(\alpha_2) = -0.1728$, $P_{-1}(\alpha_2) = 0.7407$ $P_0(\alpha_2) = -1.2963$, $P_1(\alpha_2) = 1.7284$















Lagrange Interpolation Algorithm

• A realization of the factor-of-3 interpolator in the form of a time-varying filter is shown below



24

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Arbitrary-Rate Sampling Rate Converter

• As a result, the fractional-rate sampling rate converter is almost realized in a hybrid form as indicated below for the case of an interpolator

$$x[n] \xrightarrow{f_{T}} H(z) \xrightarrow{ranalog"} y[n]$$

• The digital sampling rate converter can be implemented in a multistage form to reduce the computational complexity

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57