Infinite impulse response filters

Leakey Integrator filter

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April 2020

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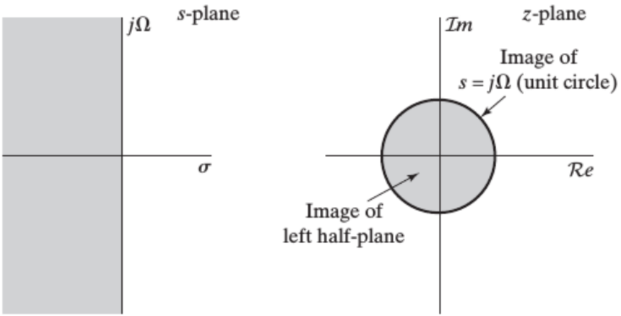
Table: Classification of discrete filters

	Finite impulse response (FIR)	Infinite impulse response (IIR)
Filtering in time domain	Moving average	Leaky Integrator
Filtering in frequency domain	Windowed Filters Equiripple Minimax	Bilinear z-transform ZOH method

- The main idea is to transform an analog filter to the discrete domain.
- From *s* domain to *z* domain.
- This way, all the theory behind analog filter can be reused to implement a filter in a computer.

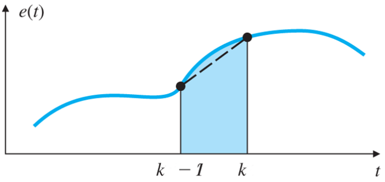


Suppose the following integrator,

$$\frac{U(s)}{E(s)} = D_c(s) = \frac{1}{s}\,. \qquad (1)$$

The area under $e(t)$ over $k \times T$ periods is,

$$u(k) = \int_0^{k-1} e(t)dt + \int_{k-1}^{k} e(t)dt\,. \qquad (2)$$



Tustin's method uses the trapezoidal integration, to approximate $e(t)$ by a straight line between two samples. The technique is an algebraic transformation between variables s and z.

$$u(k) = u(k-1) + \frac{T}{2}\left[e(k-1) + e(k)\right], \qquad (3)$$

$$U(z) = z^{-1}U(z) + \frac{T}{2}\left[z^{-1}E(z) + E(z)\right], \qquad (4)$$

$$U(z)(1-z^{-1}) = \frac{T}{2}\left[E(z)(1+z^{-1})\right], \qquad (5)$$

$$\Longrightarrow \frac{U(z)}{E(z)} = \frac{T}{2}\left(\frac{1+z^{-1}}{1-z^{-1}}\right) = \frac{1}{\frac{2}{T}\frac{1-z^{-1}}{1+z^{-1}}}\,. \qquad (6)$$

Comparing Eq. 1 and 6,

$$\boxed{s \approx \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)}\,. \qquad (7)$$

Relationship between analog and digital frequencies

- Ω is the analog frequency, $-\infty, < \Omega < \infty$.
- ω, the "digital" frequency, $-\pi, < \omega < \pi$, i.e., $-2\pi f_s/2, < \omega < 2\pi f_s/2$.
- What is the relationship between Ω and ω.

Doing $s = j\Omega$, z should be evaluated in the unity circle, so, $z = r \cdot e^{j\omega} = \cdot e^{j\omega}$, with $r = 1$.

$$s = \frac{2}{T}\left(\frac{1 - e^{-j\omega}}{1 + e^{-j\omega}}\right) = \frac{2}{T}\left[\frac{2e^{-j\omega/2}(j \sin \omega/2)}{2e^{-j\omega/2}(\cos \omega/2)}\right] = j\frac{2}{T}\tan(\omega/2)\,. \tag{8}$$

Real and imaginary parts on both sides of Eq. 8 are,

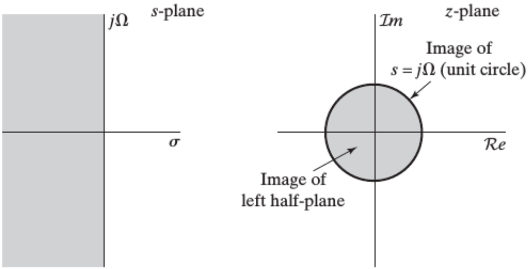
Since $s = \sigma + j\Omega$,

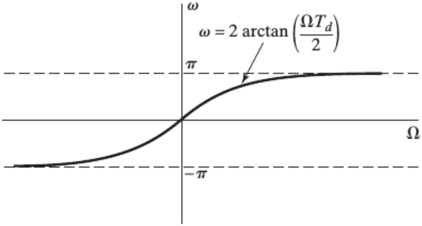
$$\sigma = 0\,, \tag{9}$$

$$\Omega = \frac{2}{T}\tan(\omega/2)\,, \tag{10}$$

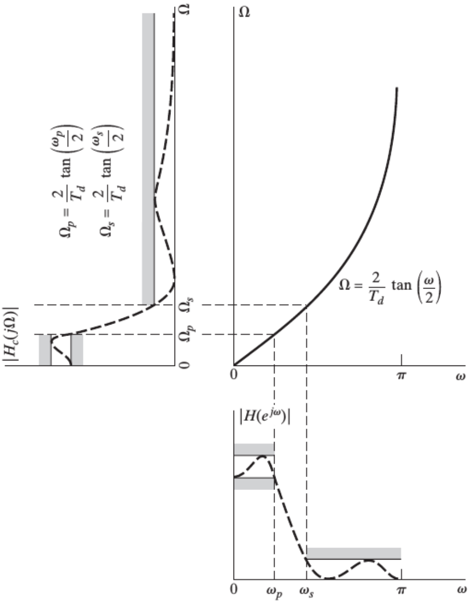
$$\Longrightarrow \quad \omega = \arctan(\Omega T/2)\,. \tag{11}$$

Map from s to z

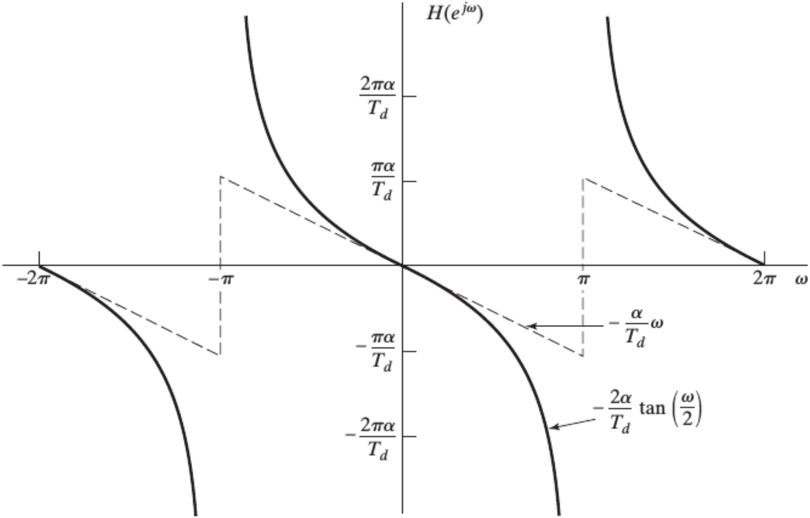




- Non-linear compression of the frequency axis.
- The design of discrete-time filters using the bilinear transformation is useful only when this compression can be tolerated or compensated for.



Suppose a continuous-time filter with linear phase response. The nonlinear warping of the frequency axis introduced by the bilinear transformation will not preserve linearity in phase response.



1 Choose the analog filter that complains with the desired performance.

For example, second-order Butterworth low-pass filter.

$$G(s) = \frac{\Omega_c^2}{s^2 + s\sqrt{2}\Omega_c + \Omega_c^2}$$

2 Cut-off digital frequency is normalized.

$$f_{dc} = 100 \text{ Hz}, \; f_s = 1000 \text{ Hz}, \; T = 0.001 \text{ s}.$$

$$\omega_c = 2\,\pi\,100/1000 = 0.628 \text{ rad/s}.$$

3 Pre-warp the analog frequencies.

$$\Omega_c = \frac{2}{T}\tan(\omega_c/2) = \frac{2}{0.001}\tan\left(\frac{0.628}{2}\right) = 649.839 \text{ rad/s}.$$

$$f_{ac} = 103.42 \text{ Hz}.$$

Example of IIR design using bilinear transform

4 Replace s by the bilinear transform, $s \approx \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)$.

$$H(s) = \frac{\Omega_c^2}{s^2 + s\sqrt{2}\Omega_c + \Omega_c^2}$$

$$H(z) = \frac{(649.84)^2}{\left(\frac{2}{T}\right)^2\left(\frac{1-z^{-1}}{1+z^{-1}}\right)^2 + \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)\sqrt{2}\,(649.84) + (649.84)^2}$$

$$H(z) = \frac{0.067 + 0.135z^{-1} + 0.067z^{-2}}{1 - 1.143z^{-1} + 0.413z^{-2}}$$

5 Invert the Z-transform to find the difference equation.

$$y[n] = 0.067\,x[n] + 0.135\,x[n-1] + 0.067\,x[n-2]1 + 1.143\,y[n-1] - 0.413\,y[n-2]$$

Direct form I IIR implementation

$$H(z) = \frac{b_0 + b_1 z^{-1} + \ldots + b_{N-1} z^{N-1}}{1 + a_1 z^{-1} + \ldots + a_{M-1} z^{M-1}}$$

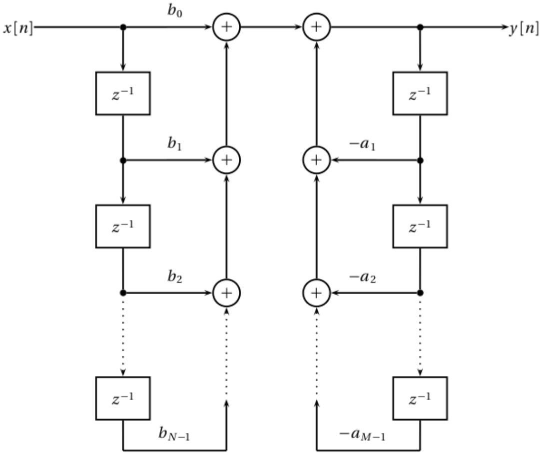


Figure 7.24 Direct Form implementation of an IIR filter.

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

By the commutative properties of the z-transform, we can invert the order of computation to turn the Direct Form I structure into a new structure.

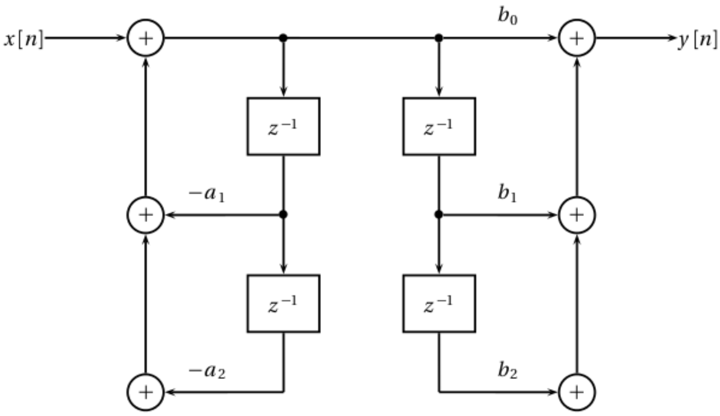


Figure 7.25 Direct form I with inverted order.

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

We can then combine the parallel delays together. This implementation is called Direct Form II; its obvious advantage is the reduced number of the required delay elements (hence of memory storage).

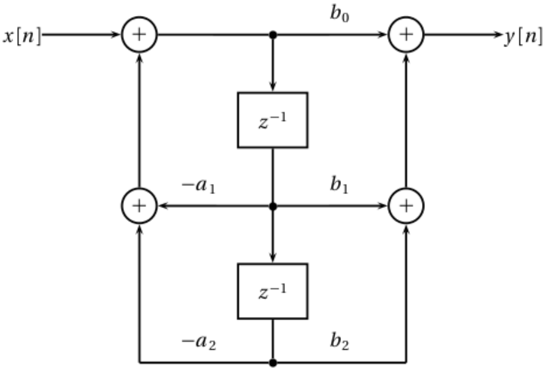


Figure 7.26 Direct Form II implementation of a second-order section.

The cascade structure of N second-order sections is much less sensitive to quantization errors than the previous Direct form II of order $2 \cdot N$.

$$H(z) = \prod_{k=1}^{N} G_k \frac{b_{0k} + b_{1k} z^{-1} + b_{2k} z^{-2}}{1 + a_{1k} z^{-1} + a_{2k} z^{-2}}$$

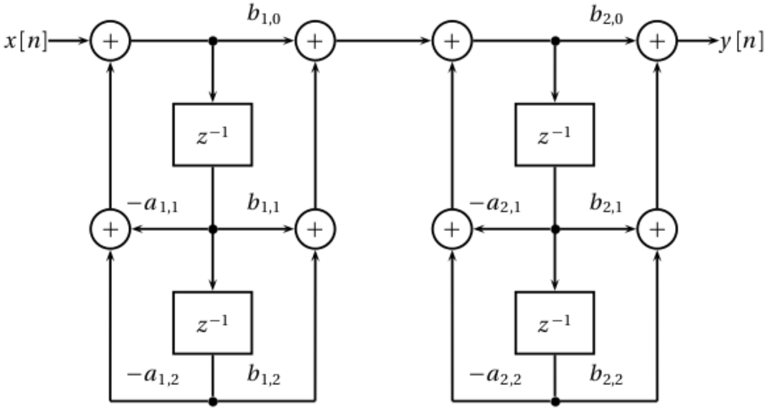


Figure 7.27 4th order IIR: cascade implementation.

FIR vs IIR

FIR, pros:

- Unconditional stability (no poles).
- Precise control of the phase response and, in particular, exact linear phase.
- Robustness with respect to finite numerical precision hardware.

FIR, cons:

- Longer input-output delay.
- Higher computational cost with respect to IIR solutions.

IIR, pros:

- Lower computational cost with respect to an FIR with similar behavior.
- Shorter input-output delay.
- Compact representation.

IIR, cons:

- Stability is not guaranteed.
- Phase response is difficult to control.
- Design is complex in the general case.
- Sensitive to numerical precision.

Bibliography

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