

Infinite impulse response filters

Bilinear z-transform

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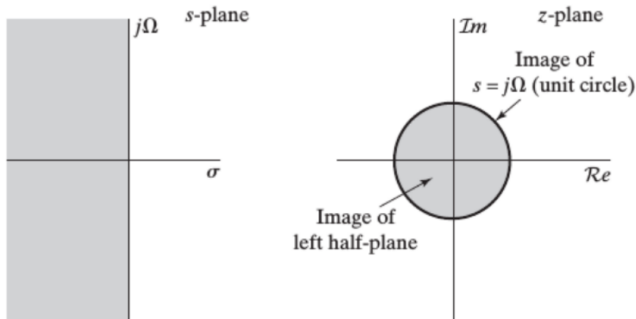
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Table: Classification of discrete filters

	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

IIR filtering in frequency domain

- 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 (Butterworth filter, Chebyshev filters, Elliptic filter).



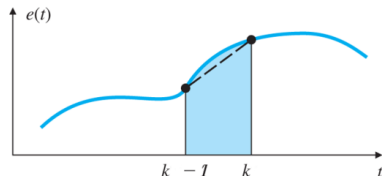
Bilinear transform (Tustin's Method)

Suppose the following integrator,

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

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

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



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 .

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

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

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

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

Comparing Eq. 1 and 6,

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

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 $r = 1$.

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

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

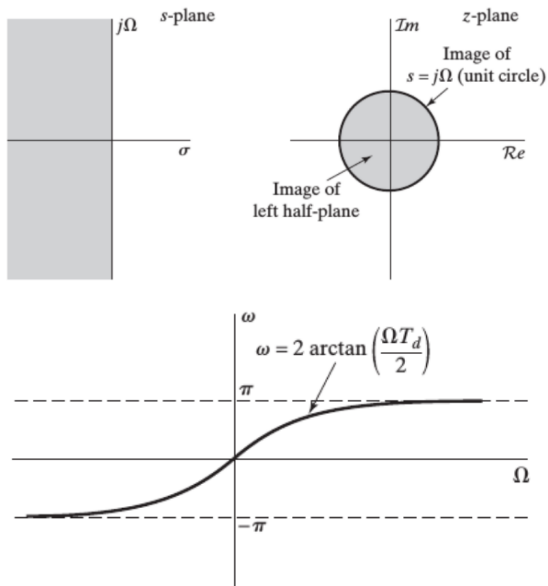
Since $s = \sigma + j\Omega$,

$$\sigma = 0, \quad (9)$$

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

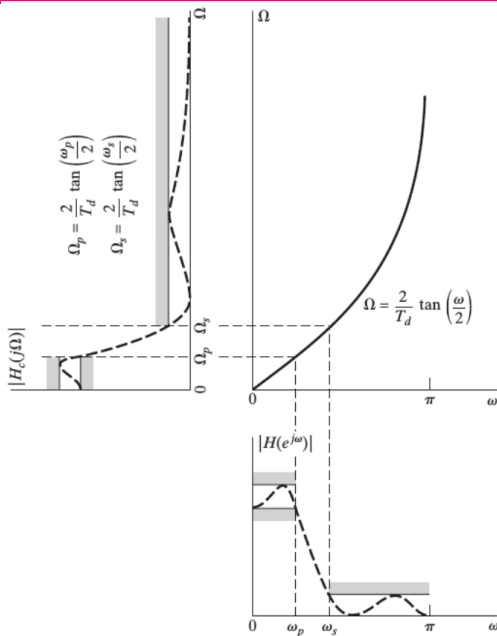
$$\implies \omega = \arctan(\Omega T/2). \quad (11)$$

Map from s to z



Frequency pre-warping

- Non-linear relationship between Ω and ω .
- Analog frequencies has to be adjust **before** analog filter design.



Example of IIR design using bilinear transform

- 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.

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

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

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

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

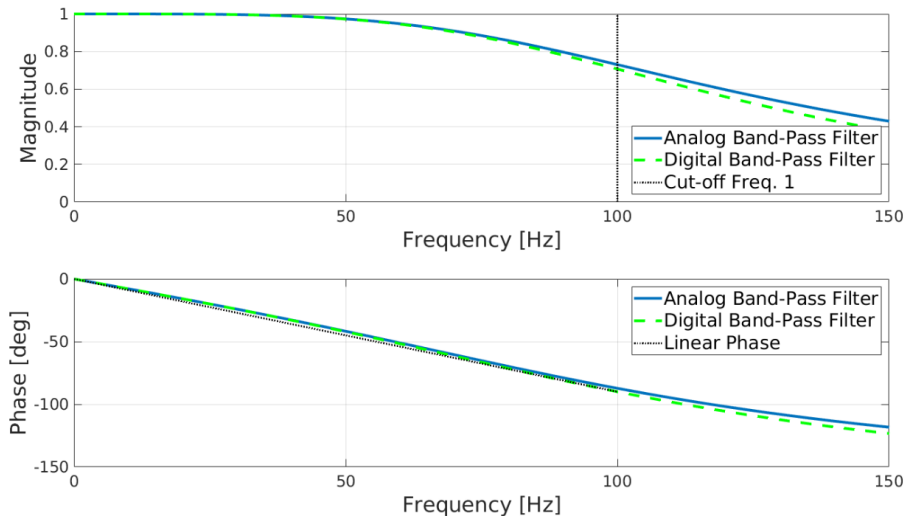
$$H(z) = \frac{(649.84)^2}{\left(\frac{2}{T}\right)^2 \left(\frac{1-z^{-1}}{1+z^{-1}}\right)^2 + \frac{2}{T} \left(\frac{1-z^{-1}}{1+z^{-1}}\right) \sqrt{2} (649.84) + (649.84)^2}$$

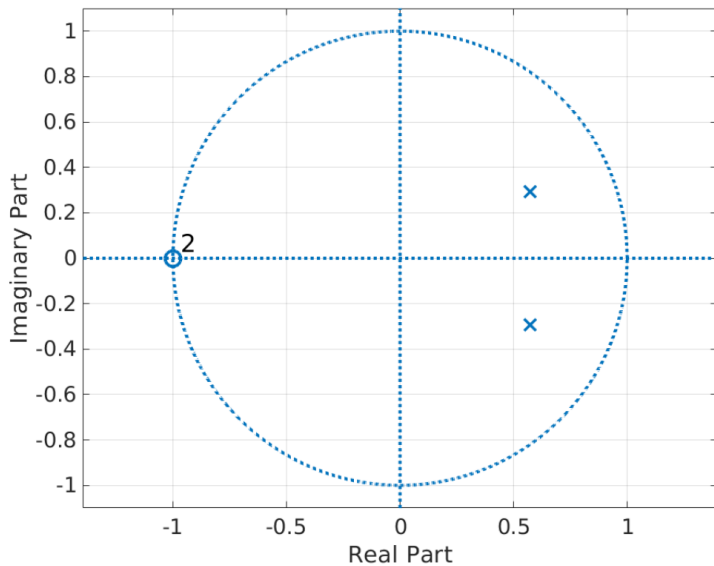
$$H(z) = \frac{0.067 + 0.135z^{-1} + 0.067z^{-2}}{1 - 1.143z^{-1} + 0.413z^{-2}}$$

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

$$\begin{aligned} y[n] = & 0.067 x[n] + 0.135 x[n-1] + 0.067 x[n-2] + \\ & + 1.143 y[n-1] - 0.413 y[n-2] \end{aligned} \quad (12)$$

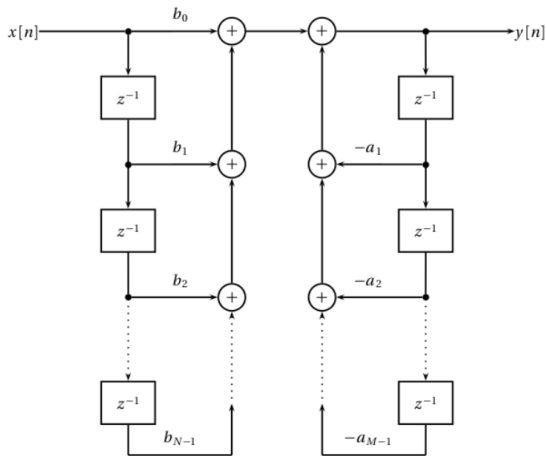
Frequency and phase responses





Direct form I IIR implementation

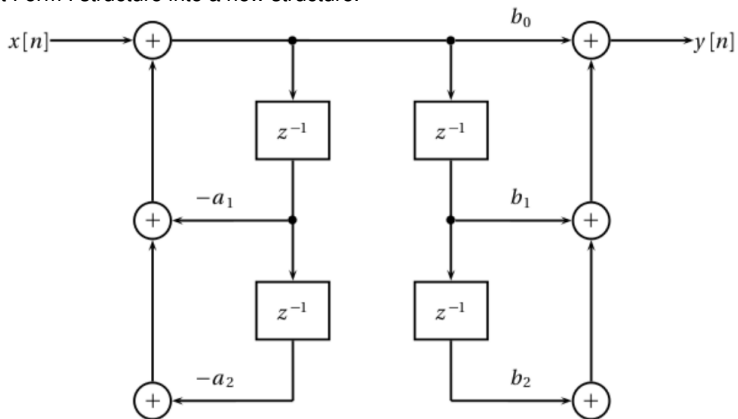
$$H(z) = \frac{b_0 + b_1 z^{-1} + \dots + b_{N-1} z^{N-1}}{1 + a_1 z^{-1} + \dots + a_{M-1} z^{M-1}}$$



Direct form I IIR implementation inverted

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

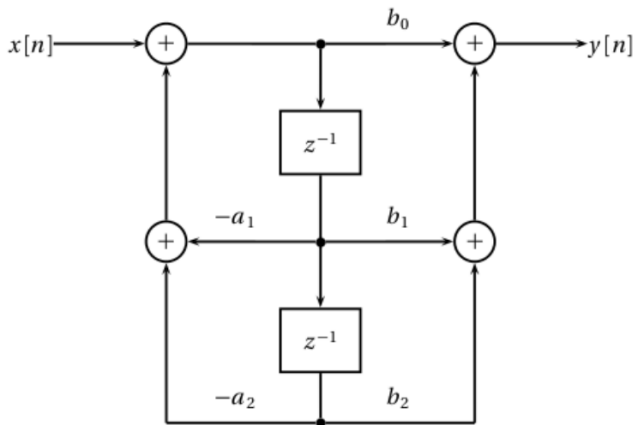
By the commutative properties of the z-transform, we can invert the order of computation to turn the Direct Form I structure into a new structure.



Direct form II IIR implementation

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

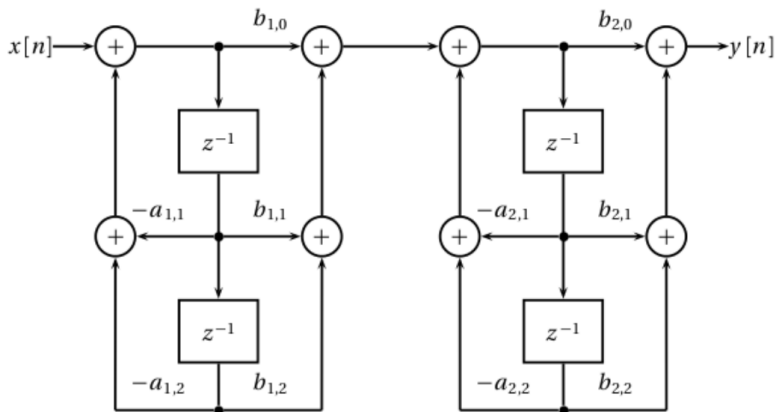
We can then combine the parallel delays together. This implementation is called Direct Form II; its obvious advantage is the reduced number of the required delay elements (hence of memory storage).



IIR cascade implementation

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

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



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.

- 1 Paolo Prandoni and Martin Vetterli. Signal processing for communications. Taylor and Francis Group, LLC. 2008. Sections 7.3, and 7.4.2.
- 2 Oliver Hinton. Digital Signal Processing Resources for EEE305 Course. Chapter 5. www.staff.ncl.ac.uk/oliver.hinton/eee305/
- 3 Gene F. Franklin, J. David Powell and Abbas Emami-Naeini. *Feedback Control of Dynamic Systems*. 7th Edition. 2014. Section 8.3.