Section M6: Filter blocks

These blocks appear at the top of the simulation area

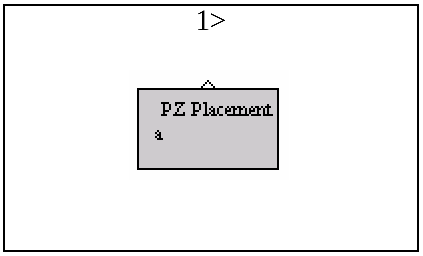
Table of blocks	
Block notation	Description
PZ-Placement	Allows entering pole/zero values
PZ-Plot	Plots poles/zeros in polar coordinates
FIR Design	FIR filter design
IIR Design	IIR filter design
Kaiser	Kaiser filter design
Parks-McClellan	Parks-McClellan filter design
LMS	LMS adaptive filter algorithm
Freq Samp.	Frequency sampling

M6.1

Block name : Pole Zero Placement **Notation**: ***PZ-Placement***

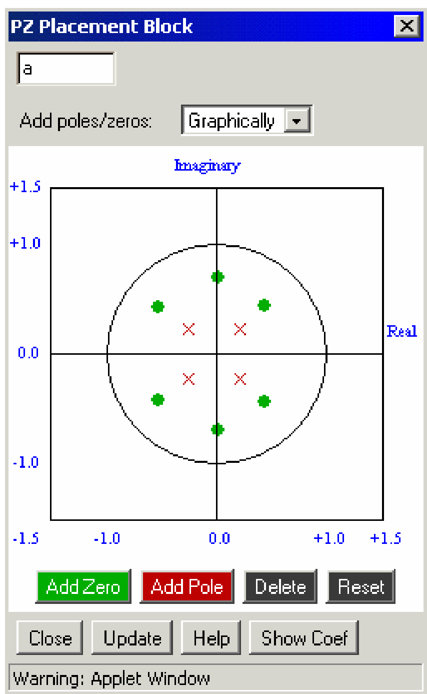
Description: This block allows the user to enter poles and zeros representing a filter. The corresponding filter coefficients are passed to the output. Poles and zeros are added as conjugate pairs, and no more than 10 (5 pairs) can be entered. They can be placed either graphically or manually. Graphical manipulation of poles and zeros is achieved through buttons that allow placing, moving and deleting. Manually placing poles and zeros can be done either in square or polar form.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)PZ-Placement dialog window

Script use:

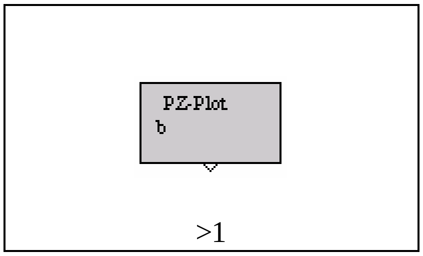
Name: pzplace

Example code: <param name = “3” value = “B3-pzplace(3,1)”>

Block name : Pole-Zero plot **Notation**: ***PZ-Plot***

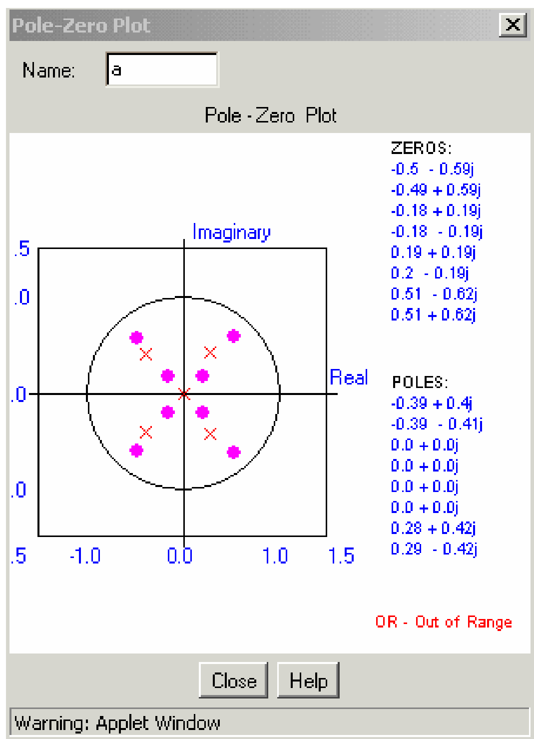
Description: This block calculates and displays the poles and zeros of a transfer function in the z-plane. The block accepts filter coefficients at its input.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)PZ-Plot dialog window

Script use:

Name: pzplot

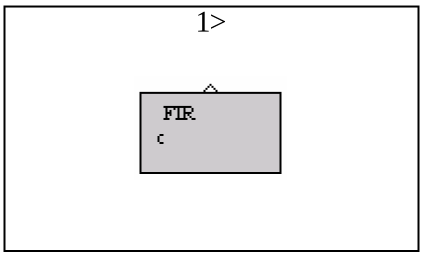
Example code: <param name = "3" value = "B3-pzplot(3,1)">

M6.3

Block name: FIR design **Notation**: ***FIR***

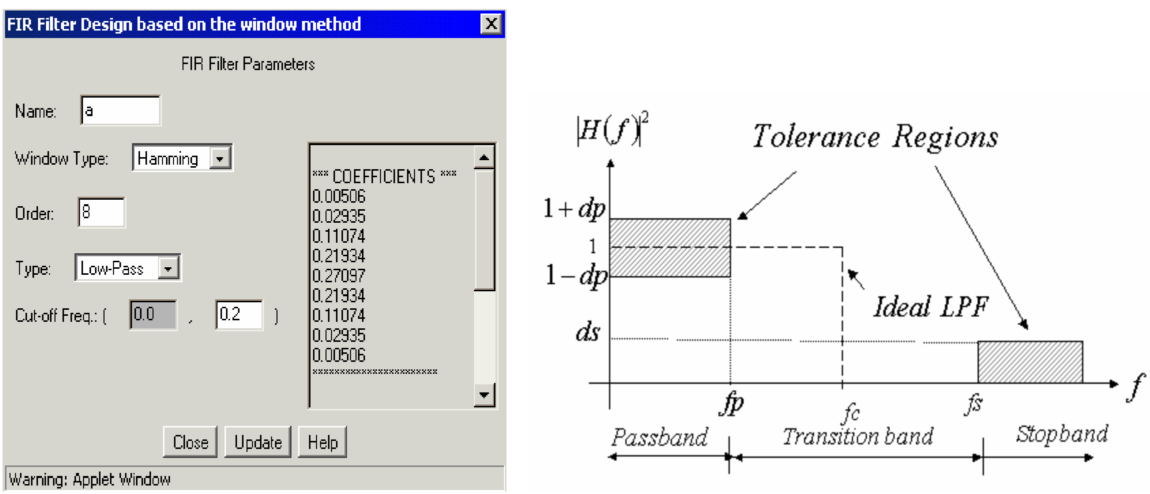
Description: Designs a finite impulse response (FIR) filter based on the windowing method. The windowing FIR filter design method is a straightforward technique implemented by expanding the frequency response of an ideal filter in a Fourier series and then truncating and smoothing the response using a window. The user needs to supply the following information: *Window type*: Hamming, Hanning, Blackman, Bartlett, rectangular or Kaiser | *Filter order* (maximum is 64) | *Type*: low-pass, high-pass, pass-band, or stop-band. Cut-off frequencies (f_c), take values from 0 to 1, where $f_c = 1$ corresponds to half-the-sampling frequency.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)FIR dialog window and filter design specifications

Script use:

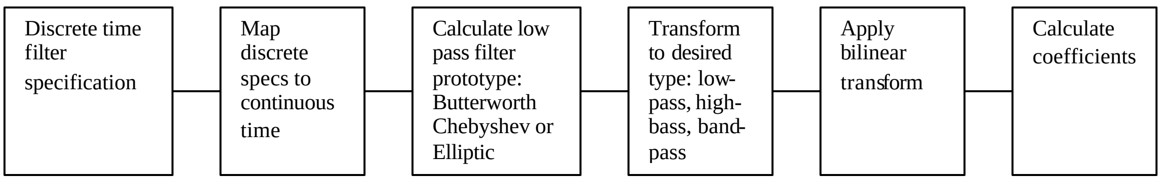
Name: FIR

Example code: <param name = “3” value = “B3-FIR(3,1)”>

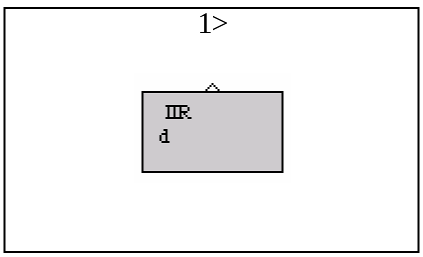
M6.4

Block name: IIR design **Notation**: ***IIR***

Description: Designs an infinite (length) impulse response (IIR) filter based on the bilinear transformation. Butterworth, Chebyshev -I & -II, and Elliptic filters are supported. The filter specifications are in terms of: *Filter typ*e- can be low-pass, high-pass or pass-band | Wp_1, Ws_1 – pass-band and stop-band edge cut-off frequencies respectively, | Wp_2, Ws_2 – second pass-band and stop-band edge cut-off frequencies respectively (for pass-band filters) | *PB*, *SB* – pass-band and stop-band tolerances in dB. Cut-off frequencies (f_c), take values from 0 to 1, where $f_c = 1$ corresponds to half-the-sampling frequency. The design process is illustrated in terms of the block diagram below:

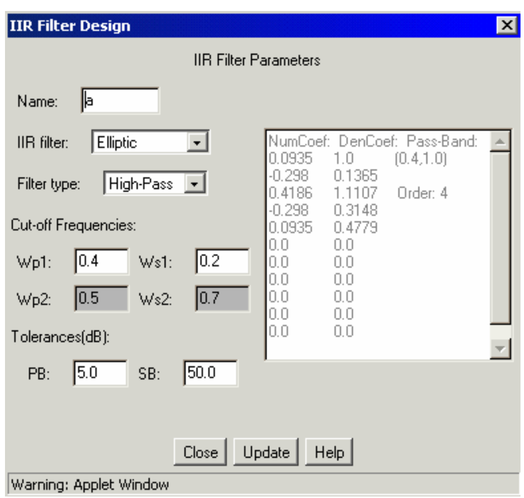


Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)FIR dialog window

Script use:

Name: IIR

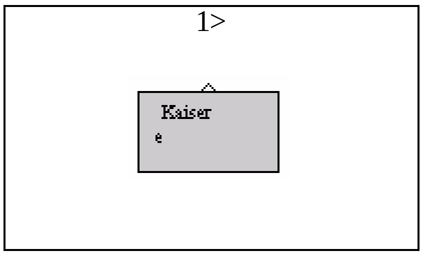
Example code: <param name = "3" value = "B3-IIR(3,1)">

M6.5

Block name : Kaiser design **Notation**: ***Kaiser***

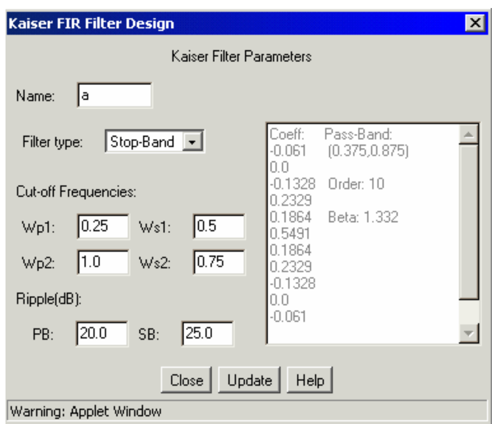
Description: This block designs Kaiser FIR filters based on the windowing method. The design process involves calculating the Fourier series of the ideal filter and then multiplying it with a Kaiser window that best fits the filter specifications. Filter specifications are: *Filter type*: can be low-pass, high-pass, stop-band or pass-band | Wp_1, Ws_1 – pass-band and stop-band edge cut-off frequencies respectively, | Wp_2, Ws_2 – second pass-band and stop-band edge cut-off frequencies respectively (for pass-band filters) | *PB*, *SB* – pass-band and stop-band tolerances in dB.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)Kaiser dialog window

Script use:

Name: Kaiser

Example code: <param name = "3" value = "B3-Kaiser(3,1)">

Equation(s) Implemented :

The order and value of b of the Kaiser window are calculated by:

$$N = \frac{A-8}{2.285\Delta w} \quad \text{and} \quad b = \begin{cases} 0.1102(A-8.7)\text{...}A > 50 \\ 0.5842(A-21)^{0.4} + 0.07886(A-21)\text{....}21 \leq A \leq 50 \\ 0\text{...}A < 21 \end{cases}$$

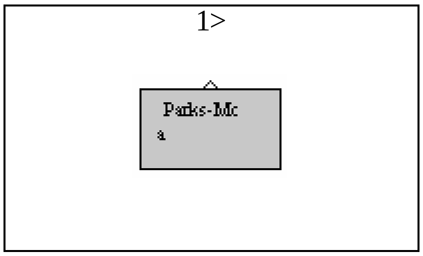
$\Delta\omega$ is the transition band of the filter and A is equal to the smaller of PB and SB.

M6.6

Block name : Parks-McClellan **Notation**: ***Parks-Mc***

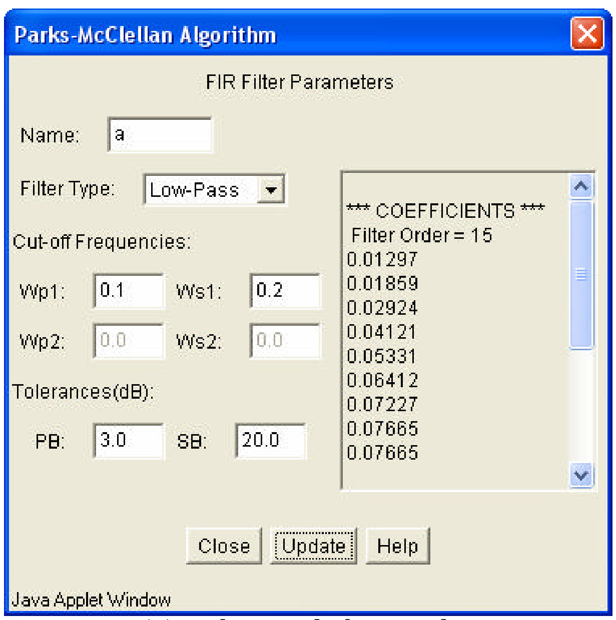
Description: This block designs FIR filters using the Parks-McClellan algorithm with min-max design. Filter specifications are: *Filter type*: can be low-pass, high-pass, stop-band, or pass-band | Wp_1, Ws_1 – pass-band and stop-band edge cut-off frequencies respectively, | Wp_2, Ws_2 – second pass-band and stop-band edge cut-off frequencies respectively (for pass-band filters) | *PB*, *SB* – pass-band and stop-band tolerances in dB.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)Parks-Mc. dialog window

Script use:

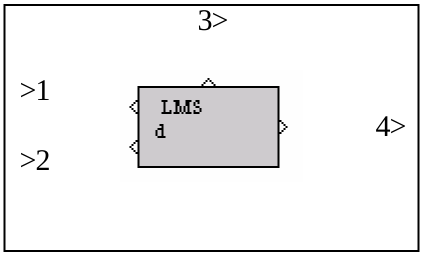
Name: ParksMac
Example code: <param name = "3" value = "B0-ParksMac(3,1)">

M6.7

Block name: Least Mean Squares Algorithm   **Notation**: ***LMS***

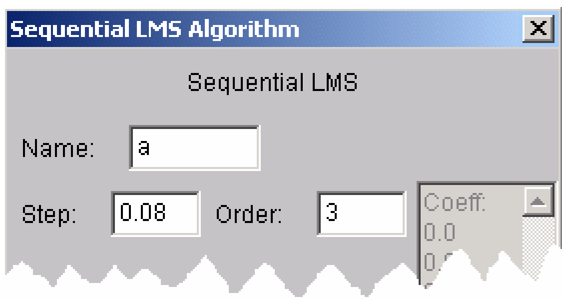
Description: Implements the sequential least mean squares adaptive filtering algorithm.

Pin assignment:



Pin	Description
1	The signal to be modeled
2	Reference signal
3	Adaptive filter coefficients
4	Adaptation error
5	
6	

Dialog window(s):



(a)LMS dialog window

Script use:

Name: LMS

Example code: <param name = “3” value = “B3-LMS(3,1)”>

Equation(s) Implemented :

A new set of adaptive filter coefficients is calculated for every new iteration in order to reduce the mean squared error. The update equation is given by

$$\boldsymbol{b}_{n+1} = \boldsymbol{b}_n + \mu e(n)\boldsymbol{x}_n$$

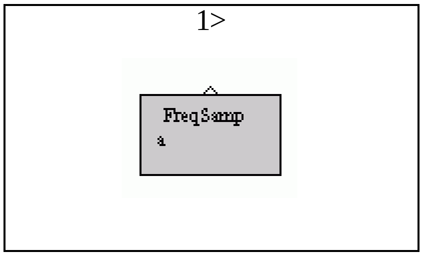
where $\mathbf{b}_n = \begin{bmatrix} b_0(n) \\ b_1(n) \\ \vdots \\ \vdots \\ b_{N-1}(n) \end{bmatrix}$ is the filter coefficient vector, $\mathbf{x}_n = \begin{bmatrix} x(n) \\ x(n-1) \\ \vdots \\ \vdots \\ x(n-N+1) \end{bmatrix}$, is the input vector and

$e(n) = d(n) - \sum_{l=0}^{N-1} b_l(n)x(n-l)$ is the error signal. The step size μ is the adaptation constant that controls the rate of convergence.

Block name: Frequency sampling **Notation**: ***FreqSamp.***

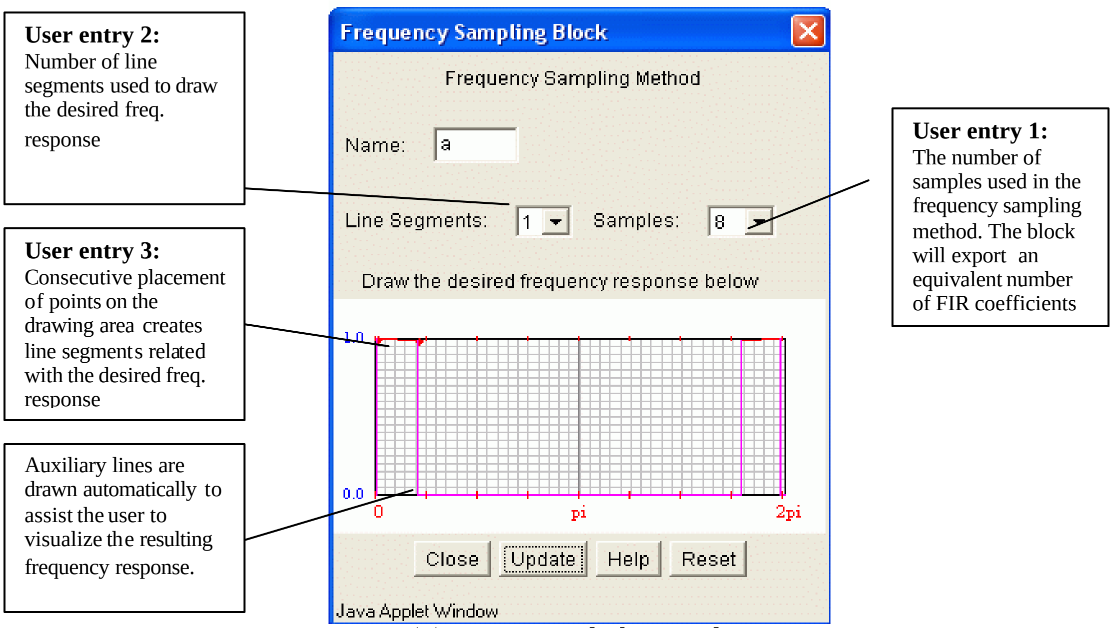
Description: This block designs a linear phase finite impulse response (FIR) filter based on the frequency sampling method. In the frequency sampling method an FIR impulse response is obtained by applying an IFFT on samples of a desired frequency response. The desired frequency response is drawn using the dialog window shown below.

Pin assignment:



Pin	Description
1	Filter coefficients
2	
3	
4	
5	
6	

Dialog window(s):



(a)FreqSamp. dialog window

Script use:

Name: FreqSamp

Example code: <param name = "3" value = "B0-FreqSamp(1,7)">

Equation(s) Implemented :

$$h(n) = \frac{1}{N}\left[\sum_{k=1}^{N/2-1} 2\left|H(k)\right|\cos\left(2\pi k(n-a)/N\right) + H(0)\right]$$

$$a = (N-1)/2, \quad k = 0, \ldots, N-1$$