Typically filters are designed by specifying frequency "bands" that define a range of frequencies and the desired response amplitude (gain) and phase in that band.

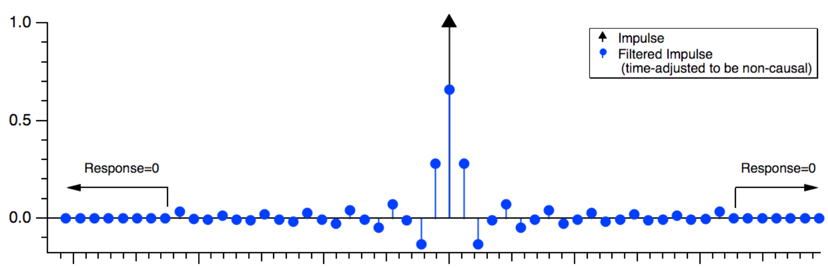
The range of frequencies that are possible range from 0 to one-half the sampling frequency of the signal, which is called the "Nyquist frequency". In the example of musicWave, the Nyquist frequency is 22,050 Hz, so filter designs for that waveform define frequency bands that end no higher than 22,050 Hz.

Filter Design Output

The result of a filter design is a set of filter "coefficients" that are used to implement the filtering. The coefficient values and format depend on the filter design type, number of bands, band frequencies, and other parameters that define the filter response. The formats for FIR and IIR designs are quite different.

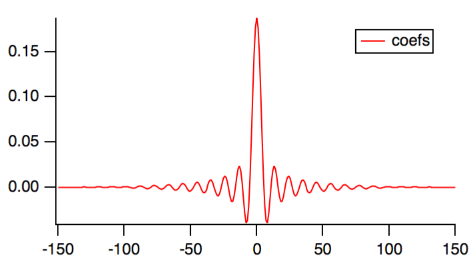
FIR Filters

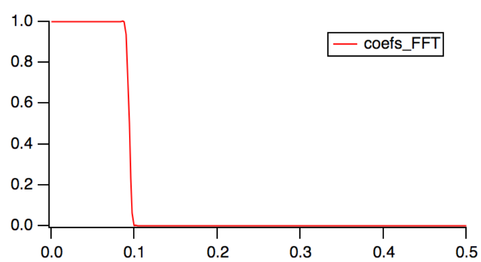
Finite Impulse Response (FIR) means that the filter's time-domain response to an impulse ("spike") is zero after a finite amount of time:



An FIR filter is a finite length, evenly-spaced time series of impulses with varying amplitudes that is convolved with an input signal to produce a filtered output signal.

The impulse response amplitudes are termed "weighting factors" or "coefficients". They are identical to the filter's response to a unit impulse. You can observe an FIR filter's frequency response by simply computing the FFT of the coefficients. If you set the X scaling of the coefficients to match the sampling frequency of the data it will be applied to, the FFT result's frequency range will be scaled to the data's Nyquist frequency. For default X scaling, the frequency range will be 0 to 0.5 Hz:



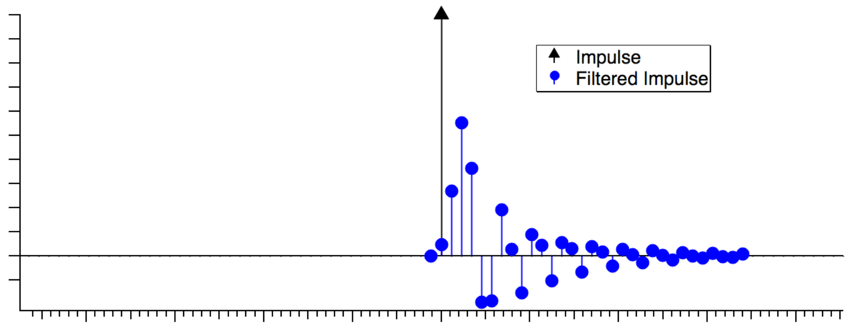


FIR filters are valued for their completely linear phase (constant delay for all frequencies), but they generally need many more coefficients than IIR filters do to achieve similar frequency responses. Consequently, electronic digital realizations of FIR filters are usually more expensive than the corresponding IIR filter.

You supply FIR coefficients to the **FilterFIR** operation along with the input waveform to compute the filtered output waveform.

IIR Filters

The response of an Infinite Impulse Response (IIR) filter continues indefinitely, as it does for analog electronic filters that employ inductors and capacitors:



An IIR filter is a set of coefficients or weights a0, a1, a2,… and b0, b1, b2… whose values and use depend on the digital implementation topology. Unlike the FIR filter, these coefficients are not the same as the filter's response to a unit impulse. See the “IIR Filter Design” topic in the “Igor Filter Design Laboratory” help file for further explanation.

IIR filters can realize quite sophisticated frequency responses with very few coefficients. The drawbacks are non-linear phase, potential for numerical instability (oscillation) when realized using limited-precision arithmetic, and the indirect design methodology (frequency transformations of conventional analog filter methods).

Igor uses two IIR implementations:

- Direct Form I (DF I)
- Cascaded Bi-Quad Direct Form II (DF II)

The IIR coefficients are represented in three forms:

- DF I
- DF II
- "zeros and poles" form

The zeros and poles form is discussed under “IIR Analog Prototype Design Graph” in the “Igor Filter Design Laboratory” help file.

You supply IIR coefficients to the FilterIIR operation along with the input waveform to compute the filtered output waveform. The format of IIR design coefficients depends on the implementation, as you can see in tables showing coefficients for Direct Form 1, Cascaded Bi-Quad Direct Form II, and pole-zero implementations of the same filter design.

Table2:coefsIIRDF1.ld

R0 Label z^0

Row	coefsIIRDF1.l	coefsIIRDF1[][(	coefsIIRDF1[][(
	x y	numerators	denominators
0	z^0	0.00386952	1
1	z^-1	0.0154781	-2.47027
2	z^-2	0.0232171	2.47797
3	z^-3	0.0154781	-1.15463
4	z^-4	0.00386952	0.208848
5			

Table3:coefsIIRDF2.ld

R0 C0 0.07874858063521141

Row	coefsIIRDF2.l	coefsIIRDF2[][0].d	coefsIIRDF2[][1].d	coefsIIRDF2[][2].d	coefsIIRDF2[][3].d	coefsIIRDF2[][4].d	coefsIIRDF2[][5].d
	secti y	a0 (numerator)	a1*z^-1	a2*z^-2	b0 (denominator)	b1*z^-1	b2*z^-2
0	section 1	0.0787486	0.157497	0.0787486	1	-1.0989	0.321633
1	section 2	0.0491376	0.0982753	0.0491376	1	-1.37137	0.649338
2							